Official Cert Guide

Study Planner

CCNA Automation

200-901

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CCNA Automation CCNAAUTO 200-901 Official Cert Guide

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Adrian Iliesiu, CCIE No. 43909 (EI), is a network engineer with over 20 years of professional experience in the field. He currently works as a Principal Engineer within the Cisco DevNet organization. Throughout his career, Adrian has held a wide range of roles, including team leader and network, systems, and QA engineer, across multiple industries and international organizations. In his current role at Cisco, Adrian focuses on advancing network programmability and automation, with particular emphasis on enterprise infrastructure. He is an established author, a distinguished speaker at Cisco Live!, and a recipient of the prestigious Cisco Pioneer Award. Adrian has also appeared on Cisco TechWise, Cisco Champion podcasts, and DevNet webinars. He is the host of the Simplifying Network Automation with NetGru series of live streams. He holds a Bachelor's degree in Electronics and Telecommunications from the Technical University of Cluj-Napoca and a Master's degree in Telecommunication Networks from Politehnica University of Bucharest.

Ashutosh Malegaonkar serves as Senior Director of Networking at NVIDIA, where he leads Developer Technology (DevTech) and Technical Marketing for NVIDIA Networking. A respected industry visionary, Ashutosh brings more than twenty-five years of pioneering experience—spanning networking platforms, AI data center networking, voice, video, search, video analytics, cloud, software, and APIs.

Before joining NVIDIA, Ashutosh dedicated twenty-three years to Cisco, earning the distinction of Distinguished Engineer—one of Cisco's most prestigious technical leadership positions. He spearheaded the DevNet Co-Creations team, partnering with key customers, partners, and developers to transform innovative ideas into impactful solutions. His diverse career also includes founding roles in two start-up ventures.

Widely regarded for his ability to inspire innovation, Ashutosh is passionate about harnessing software and APIs to address real-world customer challenges. His comprehensive expertise in networking technologies enables him to pinpoint innovative applications that deliver meaningful and measurable outcomes for customers.

Ashutosh's achievements have been recognized with numerous professional honors, including Cisco Pioneer Awards. He is a sought-after keynote and technical presenter at industry conferences such as Cisco Connect, Cisco Live, and Tech Field Day. Ashutosh holds more than 25 approved patents and is the author of two publications.

Barry Yuan, CCIE No. 11860 (R&S and SEC) is a Principal Solutions Engineer at Cisco, an inventor with many patents, as well as a Cisco Live Distinguished Speaker. He brings 25 years of experience in

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About the Technical Reviewers

This content is currently in development.

Dedications

Chris Jackson: Writing a book is a solitary effort, but the real work is shared by everyone who loves and supports you. This book is just the latest project that my beautiful wife Piper has provided infinite patience, love, and understanding as I wrote late into the evening and on weekends. She is my rock, my light, and my greatest cheerleader. My life is amazing because of her and her love. My children Caleb, Sydney, and Savannah have been so forgiving of my time commitments and allowed me to focus on delivering something I could be proud of. Each of you are so wonderful and the time away from you has been a sacrifice that I do not make lightly. Now it's time to celebrate! Last, but certainly not least, are all my friends and co-workers who take up the slack when my inability to say no to new projects rears its head. They drive me to be better, and I am fortunate to work with some of the most professional and high-quality individuals in the industry.

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Ashutosh Malegaonkar: I dedicate this second edition to my Guru, Shri Gondavlekar Maharaj, whose grace, guidance, and blessings gave me this opportunity and enabled me to see it through to fulfillment.

I dedicate this work with profound love and gratitude as a tribute to my beloved parents. My mother would have been deeply proud, and my father was always my greatest advocate, taking great pride in every one of my career accomplishments. Their wisdom and values molded the person I became and continue to guide and inspire me. I hold their memory and legacy forward with reverence and gratitude.

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Reader Services

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Practice exams: The companion website contains an exam engine that enables you to review practice exam questions. Use these to prepare with a sample exam and to pinpoint topics where you need more study.

Glossary quizzes: The companion website contains interactive quizzes that enable you to test yourself on every glossary term in the book.

Video training: The companion website contains unique test-prep videos.

To access this additional content, simply register your product. To start the registration process, go to www.ciscopress.com/register and log in or create an account.* Enter the product ISBN 9780135368084 and click Submit. After the process is complete, you will find any available bonus content under Registered Products.

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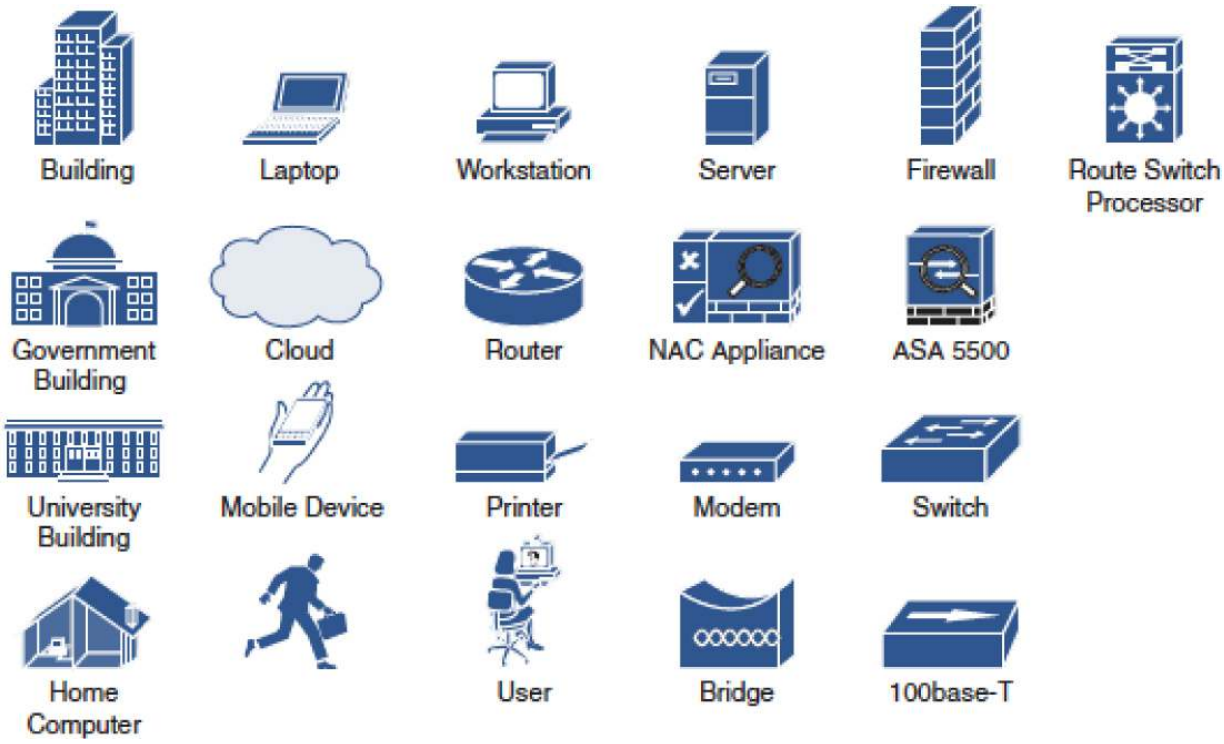
Icons Used in This Book



Command Syntax Conventions

The conventions used to present command syntax in this book are the same conventions used in the IOS Command Reference. The Command Reference describes these conventions as follows:

- **Boldface** indicates commands and keywords that are entered literally as shown. In actual configuration examples and output (not general command syntax), boldface indicates commands that are manually input by the user (such as a **show** command).
- *Italic* indicates arguments for which you supply actual values.
- Vertical bars (|) separate alternative, mutually exclusive elements.
- Square brackets ([ ]) indicate an optional element.
- Braces ({ }) indicate a required choice.
- Braces within brackets ([{ }]) indicate a required choice within an optional element.

Introduction

This book was written to help candidates improve their network programmability and automation skills—not only for preparation of the CCNAAUTO 200-901 exam, but also for real-world skills for any production environment.

Readers of this book can expect that the blueprint for the CCNAAUTO 200-901 exam tightly aligns with the topics contained in this book. This was by design. Candidates can follow along with the examples in this book by utilizing the tools and resources found on the DevNet website and other free utilities such as Postman and Python.

This book is targeted for all learners who are learning these topics for the first time, as well as for those who wish to enhance their network programmability and automation skillset.

Be sure to visit www.cisco.com to find the latest information on CCNAAUTO 200-901 exam requirements and to keep up to date on any new exams that are announced.

Goals and Methods

The most important and somewhat obvious goal of this book is to help you pass the CCNA Automation CCNAAUTO 200-901 exam. Additionally, the methods used in this book to help you pass the CCNAAUTO exam are designed to also make you much more knowledgeable about how to do your job. While this book and the companion website together have more than enough questions to help you prepare for the actual exam, the method in which they are used is not to simply make you memorize as many questions and answers as you possibly can.

One key methodology used in this book is to help you discover the exam topics that you need to review in more depth, to help you fully understand and remember those details, and to help you prove to yourself that you have retained your knowledge of those topics. So, this book does not try to help you pass by memorization but helps you truly learn and understand the topics. The CCNAAUTO exam is the foundation exam for the CCNA Automation certification, and the knowledge contained within is vitally important to consider yourself a truly skilled network developer. This book would do you a disservice if it didn't attempt to help you learn the material. To that end, the book will help you pass the CCNAAUTO exam by using the following methods:

- Helping you discover which test topics you have not mastered
- Providing explanations and information to fill in your knowledge gaps
- Supplying exercises and scenarios that enhance your ability to recall and deduce the answers to test questions

Who Should Read This Book?

This book is intended to help candidates prepare for the CCNAAUTO 200-901 exam. Not only can this book help you pass the exam, but also it can help you learn the necessary topics to provide value to your organization as a network developer.

Passing the CCNAAUTO 200-901 exam is a milestone toward becoming a better network developer. This in turn can help with becoming more confident with these technologies.

Strategies for Exam Preparation

The strategy you use for the CCNAAUTO exam might be slightly different than strategies used by other readers, mainly based on the skills, knowledge, and experience you already have obtained.

Regardless of the strategy you use or the background you have, this book is designed to help you get to the point where you can pass the exam with the least amount of time required. However, many people like to make sure that they truly know a topic and thus read over material that they already know. Several book features will help you gain the confidence that you know some material already and to also help you know what topics you need to study more.

The Companion Website for Online Content Review

All the electronic review elements, as well as other electronic components of the book, are provided on this book's companion website.

How to Access the Companion Website

To access the companion website, start by establishing a login at www.ciscopress.com and registering your book. To do so, simply go to www.ciscopress.com/register and enter the ISBN of the print book: 9780135368084. After you have registered your book, go to your account page and click the Registered Products tab. From there, click the Access Bonus Content link to get access to the book's companion website.

Note that if you buy the Premium Edition eBook and Practice Test version of this book from Cisco Press, your book will automatically be registered on your account page. Simply go to your account page, click the Registered Products tab, and select Access Bonus Content to access the book's companion website.

How to Access the Pearson Test Prep (PTP) App

You have two options for installing and using the Pearson Test Prep application: a web app and a desktop app. To use the Pearson Test Prep application, start by finding the registration code that comes with the book. You can find the code in these ways:

- **Print book:** Look in the cardboard sleeve in the back of the book for a piece of paper with your book's unique PTP code.
- **Premium Edition:** If you purchase the Premium Edition eBook and Practice Test directly from the Cisco Press website, the code will be populated on your account page after purchase. Just log in at www.ciscopress.com, click **Account** to see details of your account, and click the **Digital Purchases** tab.
- **Amazon Kindle:** For those who purchase a Kindle edition from Amazon, the access code will be supplied directly from Amazon.
- **Other Bookseller eBooks**: Note that if you purchase an eBook version from any other source, the practice test is not included because other vendors to date have not chosen to vend the required unique access code.

> **Note**
>
> Do not lose the activation code because it is the only means with which you can access the QA content with the book.

Once you have the access code, to find instructions about both the PTP web app and the desktop app, follow these steps:

Step 1. Open this book's companion website, as shown earlier in this Introduction under the heading "How to Access the Companion Website."

Step 2. Click the Practice Exams button.

Step 3. Follow the instructions listed there both for installing the desktop app and for using the web app.

If you want to use the web app only at this point, just navigate to www.pearsontestprep.com, establish a free login if you do not already have one, and register this book's practice tests using the registration code you just found. The process should take only a couple of minutes.

> **Note**
>
> Amazon eBook (Kindle) customers: It is easy to miss Amazon's email that lists your PTP access code. Soon after you purchase the Kindle eBook, Amazon should send an email. However, the email uses very generic text and makes no specific mention of PTP or practice exams. To find your code, read every email from Amazon after you purchase the book. Also do the usual checks for ensuring your email arrives, like checking your spam folder.

> **Note**
>
> Other eBook customers: As of the time of publication, only the publisher and Amazon supply PTP access codes when you purchase their eBook editions of this book.

How This Book Is Organized

Although this book can be read cover-to-cover, it is designed to be flexible and allow you to easily move between chapters and sections of chapters to cover just the material that you need more work with. Chapter 1 provides an overview of the Cisco career certifications and offers some strategies for how to prepare for the exams. The dichotomy of that is in Chapters 2 through 18. These chapters are the core chapters and can be covered in any order. If you do intend to read them all, the order in the book is an excellent sequence to use.

The core chapters, Chapters 2 through 18, cover the following topics:

- **Chapter 2, "Software Development and Design"**: This chapter introduces key software development methods, like Waterfall and Agile, and includes the common design patterns MVC and Observer. Software version control systems, how to use Git, and how to conduct code reviews are covered as well. You will also work with Bash, compare commits with diff, and walk through the pull request workflow.
- **Chapter 3, "Introduction to Python"**: This chapter provides an overview of Python syntax, working with various data types, getting input and producing output, and how to use conditionals and loops to control program flow. Type hints and the role of AI coding assistants in your exam preparation are covered as well.
- **Chapter 4, "Python Functions, Classes, and Modules"**: This chapter introduces Python functions and Object-Oriented Programming techniques. In addition, it also covers Python classes and how to work with modules to extend Python capabilities. Type hints in functions and classes are introduced as well.
- **Chapter 5, "Working with Data in Python"**: This chapter covers the various ways you can input data into your Python program, parse data, and handle errors. Finally, test-driven development is introduced as well as how to perform unit tests.
- **Chapter 6, "Application Programming Interfaces (APIs)"**: This chapter covers a high-level overview of some common API types, REST API Authentication, Simple Object Access Protocol

(SOAP), and Remote-Procedure Call (RPC) protocol as well as common examples of when and where each protocol is used.

- **Chapter 7, "RESTful API Requests and Responses"**: This chapter presents a detailed overview of REST APIs. It discusses several aspects of REST APIs including URL, methods, headers, return codes, data formats, architectural constraints, and various tools used for working with REST APIs.
- **Chapter 8, "Cisco Enterprise Networking Management Platforms and APIs"**: This chapter starts with what SDKs are and then covers Cisco Enterprise Networking Platforms and their APIs, including examples of how to interact with the APIs. The platforms covered in this chapter are Cisco Meraki, Cisco DNA Center, and Cisco SD-WAN.
- **Chapter 9, "Cisco Compute"**: This chapter introduces key Cisco Data Center and Compute Management Platforms and their associated APIs. The following platforms are covered in this chapter: Cisco ACI, Cisco UCS Manager, and Cisco Intersight. Examples of API consumption for all these platforms are also included in this chapter.
- **Chapter 10, "Cisco Collaboration Platforms and APIs"**: This chapter discusses in detail Cisco's Collaboration platforms and their associated APIs, along with examples. Platforms under consideration are Webex Teams, Cisco Finesse, Webex Meetings, Webex Devices, and Cisco Unified Call Manager.
- **Chapter 11, "Cisco Security Platforms and APIs"**: This chapter discusses in detail Cisco's Security platforms, their associated APIs along with examples. Platforms under consideration are Cisco Firepower, Cisco Umbrella, Cisco Advanced Malware Protection—AMP, Cisco Identity Services Engine—ISE, and Cisco ThreatGrid.
- **Chapter 12, "Model-Driven Programmability"**: This chapter introduces key model-driven programmability concepts and protocols. An in-depth look at YANG, YANG data models, NETCONF, RESTCONF, and Model-Driven telemetry is covered in this chapter.
- **Chapter 13, "Deploying Applications"**: This chapter covers numerous application deployment models and methods. It also introduces the core concepts of DevOps as well as an introduction to Docker and how to use it. Application security, the network services that support a deployment, and the top OWASP threats round out the chapter.
- **Chapter 14, "Application Security"**: This chapter introduces application security issues, the methods of how to secure applications via modern networking components, and various tools used. Additionally, this chapter also discusses the Open Web Application Security Project (OWASP)'s top ten.
- **Chapter 15, "Infrastructure Automation"**: This chapter introduces several infrastructure automation concepts including controller versus device-level management, infrastructure as code, continuous integration/continuous delivery pipelines, and automation tools such as Ansible, Puppet, and Chef. An overview of Cisco-related products such as Cisco NSO, Cisco VIRL, and pyATS is also presented.
- **Chapter 16, "Network Fundamentals"**: This chapter presents several key networking concepts including networking reference models, switching, and routing concepts. OSI and TCP/IP reference models, Ethernet, MAC addresses, and VLANs—as well as IPv4 and IPv6 addressing concepts—are discussed in this chapter.
- **Chapter 17, "Networking Components"**: This chapter introduces some basic networking concepts, including network definitions, types, and elements such as hubs, switches, and routers. Further, it presents and differentiates between process, fast, and CEF switching. It also introduces software-defined networking discussing management, data, and control planes.
- **Chapter 18, "IP Services"**: This chapter starts by covering several protocols and technologies that are critical to networking: DHCP, DNS, NAT, SNMP, and NTP. The chapter continues with an overview of Layer 2 versus Layer 3 network diagrams and ends with a look at how to troubleshoot application connectivity issues.

Certification Exam Topics and This Book

The questions for each certification exam are a closely guarded secret. However, we do know which topics you must know to *successfully* complete this exam. Cisco publishes them as an exam blueprint for the CCNAAUTO 200-901 exam. Table I-1 lists each exam topic listed in the blueprint along with a reference to the book chapter that covers the topic. These are the same topics you should be proficient in when working with network programmability and automation in the real world.

Table I-1 *CCNAAUTO 200-901 Exam Topics and Chapter References*

CCNAAUTO 200-901 Exam Topic	Chapter(s) in Which Topic Is Covered
1.0 Software Development and Design	2
1.1 Compare data formats (XML, JSON, and YAML)	5
1.2 Describe parsing of common data format (XML, JSON, and YAML) to Python data structures	5

1.3 Describe the concepts of test-driven development	5
1.4 Compare software development methods (agile, lean, and waterfall)	2
1.5 Explain the benefits of organizing code into methods / functions, classes, and modules	4
1.6 Explain the advantages of common design patterns (MVC and Observer)	2
1.7 Explain the advantages of version control	2
1.8 Utilize common version control operations with Git	2
1.8.a Clone	2
1.8.b Add/remove	2
1.8.c Commit	2
1.8.d Push / pull	2
1.8.e Branch	2

1.8.f Merge and handling conflicts	**2**
1.8.g diff	**2**
2.0 Understanding and Using APIs	
2.1 Construct a REST API request to accomplish a task given API documentation	**6**
2.2 Describe common usage patterns related to webhooks	**6, 7**
2.3 Describe the constraints when consuming APIs	**6**
2.4 Explain common HTTP response codes associated with REST APIs	**6**
2.5 Troubleshoot a problem given the HTTP response code, request and API documentation	**6**
2.6 Interpret the parts of an HTTP response (response code, headers, body)	**6**
2.7 Utilize common API authentication mechanisms: basic, custom token, and API keys	**6**
2.8 Compare common API styles (REST, RPC, synchronous, and asynchronous)	**6**

2.9 Construct a Python script that calls a REST API using the requests library	**7**
3.0 Cisco Platforms and Development	
3.1 Construct a Python script that uses a Cisco SDK given SDK documentation	**8, 9**
3.2 Describe the capabilities of Cisco network management platforms and APIs (Meraki, Cisco Catalyst Center, ACI, Cisco Catalyst SD-WAN, and NSO)	**8, 9, 15**
3.3 Describe the capabilities of Cisco compute management platforms and APIs (UCS Manager and Intersight)	**9**
3.4 Describe the capabilities of Cisco collaboration platforms and APIs (Webex, Webex devices, Cisco Unified Communications Manager including AXL and UDS interfaces)	**10**
3.5 Describe the capabilities of Cisco security platforms and APIs (XDR, Firepower, Secure Connect, Secure Endpoint, ISE, and Secure Malware Analytics)	**11**
3.6 Describe the device level APIs and dynamic interfaces for IOS XE and NX-OS	**12**
3.7 Describe the appropriate DevNet resource for a given scenario (Sandbox, Code Exchange, support, forums, Learning Labs, and API documentation)	**7, 8, 9, 10, 11, 12**

3.8 Apply concepts of model driven programmability (YANG, RESTCONF, and NETCONF) in a Cisco environment	**12**
3.9 Construct code to perform a specific operation based on a set of requirements and given API reference documentation such as these:	**8, 9, 15**
3.9.a Obtain a list of network devices by using Meraki, Cisco Catalyst Center, ACI, Cisco Catalyst SD-WAN, or NSO	**8, 9, 15**
3.9.b Manage spaces, participants, and messages in Webex	**10**
3.9.c Obtain a list of clients / hosts seen on a network using Meraki or Cisco Catalyst Center	**8**
4.0 Application Deployment and Security	**13**
4.1 Describe the benefits of edge computing	**13**
4.2 Describe the attributes of different application deployment models (private cloud, public cloud, hybrid cloud, and edge)	**13**
4.3 Describe the attributes of these application deployment	**13**

types	
4.3.a Virtual machines	13
4.3.b Bare metal	13
4.3.c Containers	13
4.4 Describe components for a CI/CD pipeline in application deployments	13, 15
4.5 Construct a Python unit test	5
4.6 Interpret contents of a Dockerfile	13
4.7 Utilize Docker images in local developer environment	13
4.8 Describe application security issues related to secret protection, encryption (storage and transport), and data handling	13, 14
4.9 Explain how firewall, DNS, load balancers, and reverse proxy in application deployment	13, 14
4.10 Describe top OWASP threats (such as XSS, SQL injections, and CSRF)	13, 14

4.11 Utilize Bash commands (file management, directory navigation, and environmental variables)	**2**
4.12 Describe the principles of DevOps practices	**13**
5.0 Infrastructure and Automation	**15**
5.1 Describe the value of model driven programmability for infrastructure automation	**12**
5.2 Compare controller-level to device-level management	**15**
5.3 Describe the use and roles of network simulation and test tools (such as Cisco Modeling Labs and pyATS)	**15**
5.4 Describe the components and benefits of CI/CD pipeline in infrastructure automation	**13, 15**
5.5 Describe the principles of infrastructure as code	**15**
5.6 Describe the capabilities of automation tools such as Ansible, Terraform, and Cisco NSO	**15**
5.7 Identify the workflow being automated by a Python script that uses Cisco APIs including ACI, Meraki, Cisco Catalyst Center, and RESTCONF	**8, 9**

5.8 Interpret the workflow being automated by an Ansible playbook (management packages, user management related to services, basic service configuration, and start/stop)	**15**
5.9 Interpret the workflow being automated by a bash script (such as file management, app install, user management, directory navigation)	**2**
5.10 Interpret the results of a RESTCONF or NETCONF query	**12**
5.11 Interpret basic YANG models	**12**
5.12 Interpret a unified diff	**2**
5.13 Describe the principles and benefits of a code review process	**2**
5.14 Interpret a sequence diagram that includes API calls	**7**
6.0 Network Fundamentals	
6.1 Describe the purpose and usage of MAC addresses and VLANs	**16**
6.2 Describe the purpose and usage of IP addresses, routes, subnet mask / prefix, and gateways	**16**
6.3 Describe the function of common networking components (such as switches, routers, firewalls, and load balancers)	**16, 17**
6.4 Interpret a basic network topology diagram with elements such as switches, routers, firewalls, load balancers, and port values	**16, 17**
6.5 Describe the function of management, data, and control planes in a network device	17
6.6 Describe the functionality of these IP Services: DHCP, DNS, NAT, SNMP, NTP	**18**
6.7 Recognize common protocol port values (such as, SSH, Telnet, HTTP, HTTPS, and NETCONF)	**12**
6.8 Diagnose application connectivity issues (NAT problem, Transport Port blocked, proxy, and VPN)	**18**
6.9 Explain the impacts of network constraints on applications	**18**

Each version of the exam can have topics that emphasize different functions or features, and some topics can be rather broad and generalized. The goal of this book is to provide the most comprehensive coverage to ensure that you are well prepared for the exam. Although some chapters might not address specific exam topics, they provide a foundation that is necessary for a clear understanding of important topics.

Your short-term goal might be to pass this exam, but your long-term goal should be to become a qualified network developer.

It is also important to understand that this book is a "static" reference, whereas the exam topics are dynamic. Cisco can and does change the topics covered on certification exams often.

This exam guide should not be your only reference when preparing for the certification exam. You can find a wealth of information available at Cisco.com that covers each topic in great detail. If you think that you need more detailed information on a specific topic, read the Cisco documentation that focuses on that topic.

Note that as automation technologies continue to develop, Cisco reserves the right to change the exam topics without notice. Although you can refer to the list of exam topics in Table I-1, always check Cisco.com to verify the actual list of topics to ensure that you are prepared before taking the exam. You can view the current exam topics on any current Cisco certification exam by visiting the Cisco.com website, choosing Menu, and Training & Events, then selecting from the Certifications list. Note also that, if needed, Cisco Press might post additional preparatory content on the web page associated with this book at http://www.ciscopress.com/title/9780135368084. It's a good idea to check the website a couple of weeks before taking your exam to be sure that you have up-to-date content.

Figure Credits

This content is currently in development.

Chapter 1. Introduction to the CCNA Automation Certification

This chapter covers the following topics:

- **Why Get Certified**: This section covers the benefits and advantages of becoming Cisco certified.
- **Cisco Career Certification Overview**: This section provides a high-level overview of the Cisco career certification portfolio.
- **Cisco Automation Certifications:** This section covers the CCNA Automation, CCNP Automation, Specialist, and CCIE Automation certifications, which carried DevNet names until the February 3, 2026 transition to Cisco Automation naming.
- **Cisco DevNet Overview**: This section provides an overview of DevNet, discusses the value DevNet provides to the industry, and covers the resources available and how to best leverage them.

Do I Know This Already?

The "Do I Know This Already?" quiz allows you to assess whether you should read this entire chapter thoroughly. If you are in doubt about your answers to these questions or your own assessment of your knowledge of the topics, read the entire chapter. Table 1-1 lists the major headings in this chapter and their corresponding "Do I Know This Already?" quiz questions. You can find the answers in Appendix A, "Answers to the 'Do I Know This Already?' Quiz Questions."

Table 1-1 *"Do I Know This Already?" Foundation Topics Section-to-Question Mapping*

Foundations Topic Section	Questions
Why Get Certified	2
Cisco Career Certification Overview	1, 4
Cisco Automation Certifications	3
Cisco DevNet Overview	5

Caution

The goal of self-assessment is to gauge your mastery of the topics in this chapter. If you do not know the answer to a question or are only partially sure of the answer, you should mark that question as wrong for purposes of self-assessment. Giving yourself credit for an answer that you correctly guess skews your self-assessment results and might provide you with a false sense of security.

1. Which of the following are levels of accreditation for Cisco certification? (Choose three.)

a. Associate

b. Entry

c. Authority

d. Expert

e. Maestro

2. What are some benefits certification provides for candidates? (Choose three.)

a. Highlights skills to employer

b. Increases confidence

c. Makes candidate appear smarter than peers

d. Reduces workload

e. Improves credibility

3. What type of exams are necessary to obtain CCNP Automation certification? (Choose two.)

a. Technology Core exam

b. Lab exam

c. CCT

d. Expert-level written exam

e. Concentration exam

4. True or false: In the new certification model, only a single exam is required to become CCNA certified.

a. True

b. False

5. Which of the following are part of DevNet? (Choose all that apply.)

a. Community

b. Technologies

c. Events

d. Cisco Automation Platform

e. Support

Foundation Topics

Why Get Certified

The IT industry is constantly changing and evolving. As time goes on, an ever-increasing number of technologies are putting strain on networks. New paradigms are formed as others fall out of favor. New advances are being developed and adopted in the networking realm. These advances provide faster innovation and the ability to adopt relevant technologies in a simplified way. We therefore need more intelligence and the capability to leverage the data from connected and distributed environments such as the campus, branch, data center, and WAN. Data is being used in interesting and more powerful ways than ever in the past. The following are some of the advances driving these outcomes:

• Artificial intelligence (AI)

- Machine learning (ML)
- Cloud services
- Virtualization
- Internet of Things (IoT)

The influx of these technologies is putting strain on IT operations staff, who are required to do more robust planning, find relevant use cases, and provide detailed adoption journey materials for easy consumption. All these requirements are becoming critical to success. Another area of importance is the deployment and day-to-day operations of these technologies as well as how they fit within the network environment. Some of these technologies tend to disrupt typical operations and present challenges in terms of how they will be consumed by the business. Some advances in technology are being adopted to reduce cost of operations as well as complexity. It can be said that every network, to some degree, has inherent complexity. Having tools to help manage this burden is becoming a necessity.

Many in the industry are striving for automation to handle networks as they become more and more complicated. Businesses are often forced to operate with lean IT staffs and flat or shrinking budgets; they must struggle to find ways to increase the output of what the network can do for the business. Another driver for the adoption of these technologies is improving the overall user experience within the environment. Users often need the flexibility and capability to access any business-critical application from anywhere in the network and want to have an exceptional experience. In addition to trying to improve user experience, operations staff seek ways to simplify the operations of the network. There are many inherent risks associated with manually configuring networks. One risk is not being able to move fast enough when deploying new applications or services to the network. In addition, misconfigurations can cause outages or suboptimal network performance, which can impact business operations and potentially cause financial repercussions. Finally, there is risk in that a business relies on its network for business-critical services but those services might not be available due to the IT operations staff not being able to scale to keep up with the demands of the business.

Cisco continues to report that up to 95% of network changes are performed manually. Manual changes can introduce typos, improperly used commands, and configuration inconsistencies, which can affect reliability in the network environment. Dealing with such issues is where automation shines. Automation makes it possible to signify the intent of a change that needs to be made, such as deploying quality of service across the network, and then having the network configure it properly and automatically. Automation can configure services or features with great speed and is a tremendous value to a business. Simplifying operations while reducing human error can ultimately reduce risk and potentially lower complexity.

As a simple analogy, think of an automobile. The reason most people use an automobile is to meet a specific desired outcome: to get from point A to point B. An automobile is operated as a holistic system, not as a collection of parts that make up that system. For example, the dashboard provides the user all the necessary information about how the vehicle is operating and the current state of the vehicle. To use an automobile, a driver must take certain operational steps, such as putting it in gear and then using the system to get from point A to point B. Figure 1-1 illustrates this analogy.

Figure 1-1 *Automobile as a System*

We can think of networks as systems much as we think of automobiles as systems. For many decades, the industry has thought of a network as a collection of devices such as routers, switches, and wireless

components. The shift in mindset to look at a network as a holistic system is a more recent concept that stems from the advent of network controllers, which split role and functionality from one another. This is often referred to as separating the control plane from the data plane. At a high level, the control plane is where all the instructions on a device live (for example, the routing protocols that exchange routing updates). The data plane is where all the user or data traffic flows (for example, the traffic between users on a network). Having a controller that sits on top of the rest of the devices makes it possible to operate the network as a whole from a centralized management point—much like operating an automobile from the driver's seat rather than trying to manage the automobile from all the pieces and components of which it is composed. To put this in more familiar terms, think of the command-line interface (CLI). The CLI was not designed to make massive-scale configuration changes to multiple devices at the same time. Traditional methods of managing and maintaining the network aren't sufficient to keep up with the pace and demands of the networks of today. Operations staff need to be able to move faster and simplify all the operations and configurations that have traditionally gone into networking. Software-defined networking (SDN) and controller capabilities are becoming areas of focus in the industry, and they are evolving to a point where they can address the challenges faced by IT operations teams. Controllers offer the ability to manage a network as a system, which means the policy management can be automated and abstracted. They provide the capability of supporting dynamic policy changes rather than requiring manual policy changes and device-by-device configurations when something requires a change within the environment.

It is important from a career and skills perspective to adapt to the changes within the industry. Keeping on top of new skillsets is critical to maintaining relevance in the industry or job market. Becoming Cisco certified helps with this for multiple reasons, including the following:

- Highlighting skills to employers
- Highlighting skills to industry peers
- Providing value to employers
- Providing credibility
- Providing a baseline of understanding
- Building confidence
- Enabling career advancement
- Increasing salary

Cisco Career Certification Overview

Cisco has evolved the way it offers certifications. In the past, there were many separate tracks for each discipline, such as Routing and Switching, Collaboration, Service Provider, Data Center, and Security. Although there are still separate disciplines, the number of disciplines has been greatly reduced, and the process candidates go through to achieve those certifications has changed significantly. The traditional path for Routing and Switching was Cisco Certified Network Associate (CCNA), Cisco Certified Network Professional (CCNP), and then Cisco Certified Internetwork Expert (CCIE). In the past, in order to become CCNP certified, a candidate would have had to complete the CCNA certification and be in current certified status prior to completing the CCNP. In addition, for the CCIE, a written qualification exam had to be completed prior to attempting the CCIE lab exam. However, having CCNA or CCNP certification was not necessary in order to pursue the CCIE certification. Today, the certification process has been greatly simplified. As mentioned previously, Cisco has evolved the certification structure and the prerequisites for each track.

> **Note**
>
> Cisco announced on May 27, 2025 that, effective February 3, 2026, the public certification branding for this track became CCNA Automation, CCNP Automation, and CCIE Automation. This edition uses

the current Automation names, keeping the legacy DevNet labels only where they help map to the original DEVASC content and exam history.

There are five levels of accreditation for Cisco career certifications. The following sections cover how the process has evolved and ultimately created the ability for candidates to "choose their own destiny" when it comes to what certifications and skillsets they want to pursue. Figure 1-2 shows the pyramid hierarchy used to describe the five levels of accreditation in Cisco certifications:

- Architect
- Expert
- Professional
- Associate
- Entry

Figure 1-2 *Cisco Career Certification Levels of Accreditation*

Each level of accreditation has multiple certifications and exams that pertain to that specific level. The higher the level, the more skill and rigorous hands-on experience are required. For example, the CCIE lab exam is experience based, and completing it requires hands-on expertise. In addition to the five tiers of the pyramid, there are also specialist certifications that candidates can achieve in specific technologies to showcase their knowledge and base level of understanding. (These specialist certifications are covered later in this chapter.) Simplifying the certification portfolio reduced the number of certifications available in general and also improved the process of achieving these certifications. Table 1-2 lists some of the

certifications and tracks that were available in the past, as they relate to the five-level pyramid. You can see that there were a tremendous number of options available for each track.

Table 1-2 *Cisco Career Certifications Tracks Prior to Restructuring*

Entry	Associate	Professional	Expert	Architect
Cisco Certified Entry Networking Technician (CCENT)	Cisco Certified Design Associate (CCDA)	Cisco Certified Design Professional (CCDP)	Cisco Certified Design Expert (CCDE)	Cisco Certified Architect (CCAr)
Cisco Certified Technician (CCT)	CCNA Cloud	CCNP Cloud		
	CCNA Collaboration	CCNP Collaboration	CCIE Collaboration	
	CCNA Cyber Ops			
	CCNA Data Center	CCNP Data Center	CCIE Data Center	
	CCNA Industrial			
	CCNA Routing and Switching	CCNP Routing and Switching	CCIE Routing and Switching	
	CCNA Security	CCNP Security	CCIE Security	
	CCNA Service Provider	CCNP Service Provider	CCIE Service Provider	
	CCNA Wireless	CCNP Wireless	CCIE Wireless	

Table 1-3 shows the restructuring-era certification portfolio and how those certifications fit into each level of the pyramid. It remains useful as a historical snapshot, even though Cisco's public certification catalog has continued to evolve since then.

Table 1-3 *Cisco Career Certifications Tracks After Restructuring (Historical Snapshot)*

Entry	Associate	Professional	Expert	Architect
			Cisco Certified Design Expert (CCDE)	Cisco Certified Architect (CCAr)
	DevNet Associate	DevNet Professional	DevNet Expert	
Cisco Certified Technician (CCT)	CCNA	CCNP Enterprise	CCIE Enterprise Infrastructure CCIE Enterprise Wireless	
		CCNP Collaboration	CCIE Collaboration	
		CCNP Data Center	CCIE Data Center	
		CCNP Security	CCIE Security	
		CCNP Service Provider	CCIE Service Provider	

As changes were being made in the certification portfolio, several certifications were removed or consolidated. Table 1-3 reflects that restructuring period, but it should not be read as the current catalog. Cisco now offers multiple associate-level certifications, including CCNA, CCNA Cybersecurity, and CCNA Automation.

The CCNP has changed significantly as well. Previously, for example, the CCNP Routing and Switching exam consisted of three exams:

- 300-101 ROUTE
- 300-115 SWITCH
- 300-135 TSHOOT

A candidate would have to successfully pass all three of these exams as well as the CCNA in the same track in order to become CCNP Routing and Switching certified. Today, only two exams are required in order to become CCNP Enterprise certified. Candidates can now start wherever they want; there are no prerequisites, and a candidate can start earning any level of certification—even Associate, Specialist, Professional, or Expert level certification. For the CCNP Enterprise certification, the first exam is the 300-401 ENCOR exam, which covers core technologies in enterprise networks, including the following:

- Dual-stack (IPv4 and IPv6) architecture
- Virtualization
- Infrastructure
- Network assurance
- Security
- Automation

Once the ENCOR exam is completed, a concentration exam must be taken. This is perhaps the most important and fundamental change made to the CCNP. The available concentration exams include a variety of different technology specialties and allow candidates to build their own certification (or "choose their own destiny"). Each CCNP track has its own core exam and concentrations.

Cisco made a number of changes to the specialist certifications, which allow candidates to get certified in specific areas of expertise. For example, a candidate who is proficient at Cisco Firepower can pursue a specialist certification for Firepower. The specialist certifications are important because candidates, especially consultants, often have to use many different technologies in many different customer environments. Specialist certifications can help show a candidate's ability to work on a variety of projects. They also help build credibility on a plethora of different platforms and technologies. For example, a candidate looking to focus on routing and Cisco SD-WAN could take the CCNP 300-401 ENCOR exam and then take the 300-415 ENSDWI concentration exam to become a CCNP with an SD-WAN specialty. In essence, the concentration exams are the new specialist exams, and a candidate can simply take a single specialist exam and become certified in that technology (for example, the 300-710 SNCF exam for certification in network security and Firepower).

Table 1-4 lists and describes the CCNP core, concentration, and specialist exams that were available when this edition was assembled. Read it as a historical snapshot rather than a live catalog.

Table 1-4 *CCNP Core and Concentration Exams (Historical Snapshot)*

Track	Description	Concentration Exam
Enterprise	Cisco Certified Specialist–Enterprise Core	300-401 ENCOR
Enterprise	Cisco Certified Specialist–Enterprise Advanced Infrastructure Implementation	300-410 ENARSI
Enterprise	Cisco Certified Specialist–Enterprise SD-WAN Implementation	300-415 ENSDWI
Enterprise	Cisco Certified Specialist–Enterprise Design	300-420 ENSLD
Enterprise	Cisco Certified Specialist–Enterprise Wireless Design	300-425 ENWLSD
Enterprise	Cisco Certified Specialist–Enterprise Wireless Implementation	300-430 ENWLSI
Data Center	Cisco Certified Specialist–Data Center Core	300-601 DCCOR
Data Center	Cisco Certified Specialist–Data Center Design	300-610 DCID
Data Center	Cisco Certified Specialist–Data Center Operations	300-615 DCIT
Data Center	Cisco Certified Specialist–Data Center ACI Implementation	300-620 DCACI
Data Center	Cisco Certified Specialist–Data Center SAN Implementation	300-625 DCSAN
Security	Cisco Certified Specialist–Security Core	300-701 SCOR
Security	Cisco Certified Specialist–Network Security Firepower	300-710 SNCF

Security	Cisco Certified Specialist–Network Security VPN Implementation	300-730 SVPN
Security	Cisco Certified Specialist–Email Content Security	300-720 SESA
Security	Cisco Certified Specialist–Web Content Security	300-725 SWSA
Security	Cisco Certified Specialist–Security Identity Management Implementation	300-715 SISE
Service Provider	Cisco Certified Specialist–Service Provider Core	300-501 SPCOR
Service Provider	Cisco Certified Specialist–Service Provider Advanced Routing Implementation	300-510 SPRI
Service Provider	Cisco Certified Specialist–Service Provider VPN Services Implementation	300-515 SPVI
Collaboration	Cisco Certified Specialist–Collaboration Core	300-801 CLCOR
Collaboration	Cisco Certified Specialist–Collaboration Applications Implementation	300-810 CLICA
Collaboration	Cisco Certified Specialist–Collaboration Call Control & Mobility Implementation	300-815 CLACCM
Collaboration	Cisco Certified Specialist–Collaboration Cloud & Edge Implementation	300-820 CLCEI

Note

The exams listed in Table 1-4 were available at the time of publication and are kept here for historical reference. Please visit https://www.cisco.com/go/certifications to see the current certification catalog and associated tracks.

In addition to the Associate- and Professional-level certifications, the Cisco certified specialist certifications have changed as well. Previously, in some cases multiple exams had to be completed to become certified as a specialist in a specific topic or discipline. With the new changes, however, candidates can take and complete any one of the specialist exams mentioned in Table 1-4 to become certified in that technology area. For example, a candidate who is proficient with Cisco Identity Services Engine (ISE) could pursue a specialist certification for security identity management implementation by taking the 300-715 SISE exam.

Another major change to the certification program was the reshaping of the flagship CCIE program during the restructuring period. The restructuring-era expert tracks included the following:

- Cisco Certified Design Expert (CCDE)
- CCIE Enterprise Infrastructure
- CCIE Enterprise Wireless
- CCIE Collaboration
- CCIE Data Center
- CCIE Security
- CCIE Service Provider

Current Cisco materials also include CCIE Automation. Cisco also announced a dedicated Wireless track (CCNP Wireless and CCIE Wireless) on August 26, 2025, with the new wireless exams available for testing beginning March 19, 2026.

Cisco Automation Certifications

The following sections provide an overview of the Cisco Automation certifications (CCNA Automation, CCNP Automation, and CCIE Automation) that anchor the 200-901 blueprint this book covers. These certifications carried DevNet names until February 3, 2026, so the legacy names are noted where they help you map older Cisco content to the current certification pages.

> **Note**
>
> Cisco has since announced the Expert level of this track, which is now CCIE Automation. Earning it requires passing 350-901 AUTOCOR and the CCIE Automation practical exam. When you review newer Cisco content, match certifications by exam number as well as by name.

CCNA Automation Certification

Considering everything covered up to this point in the chapter and the main focus of this book, this section covers the CCNA Automation certification at a high level. The most important thing to remember is that this is the same certification: the 200-901 blueprint this book covers is now the CCNA Automation certification. The exam number itself carries forward: on February 3, 2026, the 200-901 DEVASC exam was renamed 200-901 CCNAAUTO (Automating Networks Using Cisco Platforms), and active DevNet Associate certifications were automatically recognized as CCNA Automation.

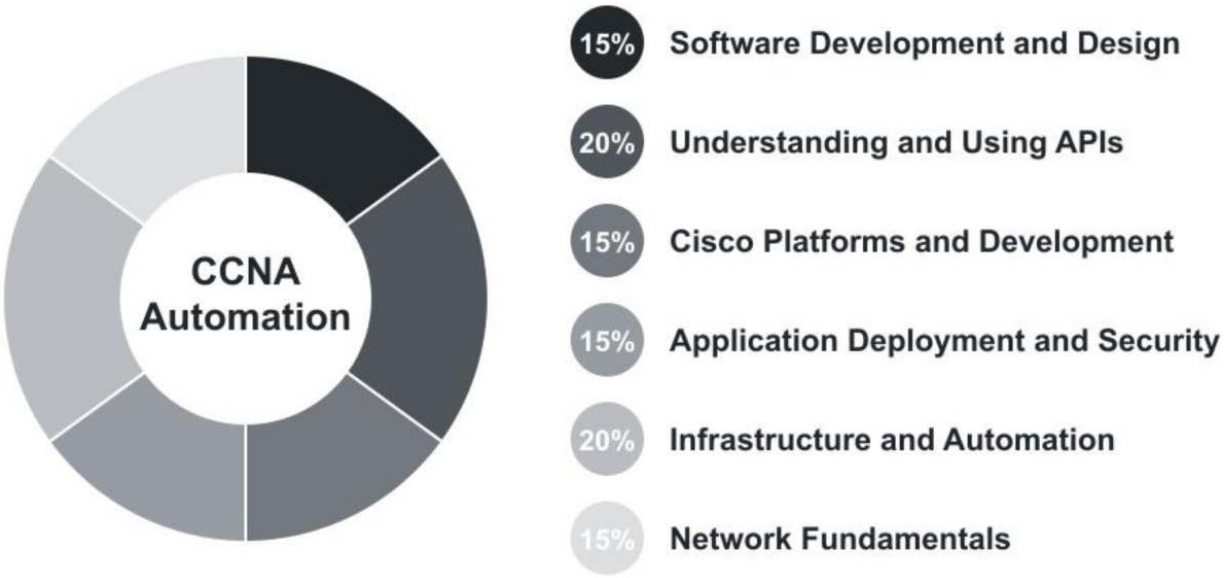


Figure 1-3 *CCNA Automation (200-901 CCNAAUTO v1.1) Knowledge Domains*

It is recommended that candidates attempting the CCNA Automation exam have at least one year of experience developing and maintaining applications built on top of Cisco platforms. In addition, they must have hands-on experience with programming languages such as Python. This certification was designed for early-in-career developers and for experienced network engineers looking to expand their skillsets to include software and automation practices. It is important to note that the line is blurring between network engineers and developers. The two skillsets are slowly merging, and candidates are becoming "network developers." Having a certification like the CCNA Automation can open doors for candidates to approach new job roles that didn't necessarily exist in the past including the following:

- Junior or entry-level DevOps engineer
- Cloud developer
- Automation engineer

When pursuing any certification, a candidate should remember the reason the certification is important in the first place. Certifications can help expand a candidate's current skillset as well as ensure a baseline level of knowledge around specific topics. The CCNA Automation certification can help businesses find individuals who possess a certain level of programmability or automation skills. It gives businesses a clear way to determine the base level of skills when looking at hiring a candidate and ensure that new hires have the necessary relevant skills. The upcoming chapters of this book align directly with the CCNA Automation blueprint. This book covers the topics necessary to build a foundational level of understanding for candidates to feel comfortable in pursuing the CCNA Automation certification.

CCNP Automation Certification

The next certification in the path after CCNA Automation is CCNP Automation. Candidates earn it by passing 350-901 AUTOCOR and either 300-435 ENAUTO or 300-635 DCNAUTO. This more robust certification requires a more advanced skillset and maps naturally to the broader Automation journey Cisco now highlights in its certification portfolio. Figure 1-4 illustrates the high-level requirements and topic domains for that next step.

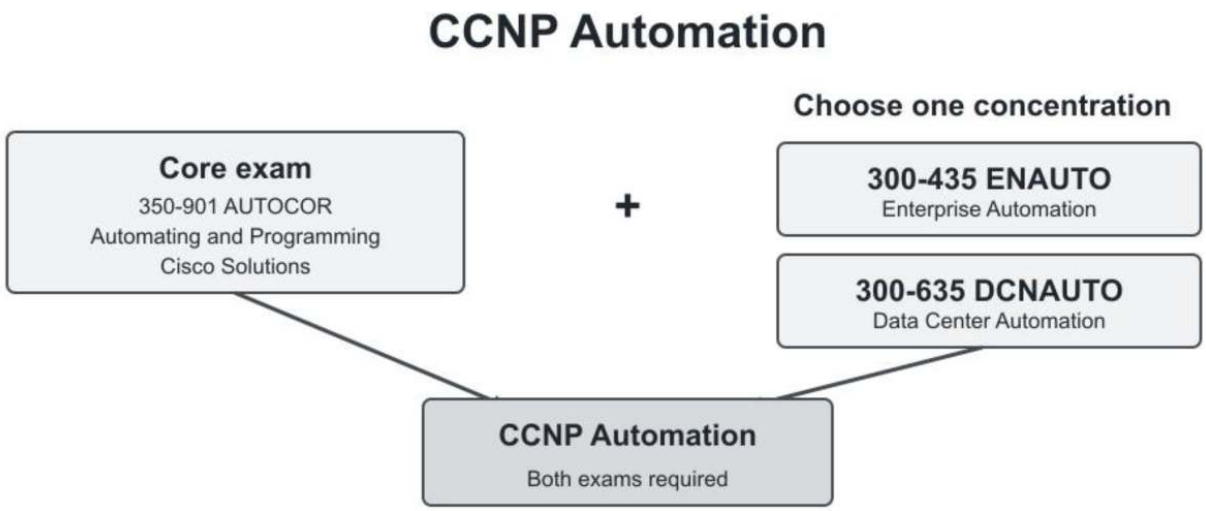


Figure 1-4 *CCNP Automation Certification Requirements*

It is recommended that candidates attempting the CCNP Automation exams have a minimum of three to five years of experience designing and implementing applications built on top of Cisco platforms. It is also critical that they have hands-on experience with programming languages such as Python. This certification was designed for experienced network engineers looking to expand their capabilities and include software and automation on their resume. It is also designed for developers moving into automation and DevOps roles as well as for solution architects who leverage the Cisco ecosystem. Infrastructure developers designing hardened production environments will also benefit from the CCNP Automation certification. Because the CCNP Automation certification provides many avenues for a candidate to create a unique journey, it is integral to aligning candidates' skillsets with their daily job tasks.

Table 1-5 lists the DevNet concentration and specialist exams that were current for this book's original certification framing. Treat it as a legacy snapshot rather than a live list of currently available exams.

Table 1-5 *DevNet Concentration and Specialist Exams (Legacy Snapshot)*

Track	Description	Specialist Exam
DevNet, Enterprise	Cisco Certified DevNet Specialist–Enterprise Automation and Programmability	300-435 ENAUTO
DevNet, Data Center	Cisco Certified DevNet Specialist–Data Center Automation and Programmability	300-635 DCAUTO
DevNet, Security	Cisco Certified DevNet Specialist–Security Automation and Programmability	300-735 SAUTO
DevNet, Service Provider	Cisco Certified DevNet Specialist–Service Provider Automation and Programmability	300-535 SPAUTO
DevNet, Collaboration	Cisco Certified DevNet Specialist–Collaboration Automation and Programmability	300-835 CLAUTO
DevNet	Cisco Certified DevNet Specialist–Core	300-901 DEVCOR
DevNet	Cisco Certified DevNet Specialist–DevOps	300-910 DEVOPS
DevNet	Cisco Certified DevNet Specialist–IoT	300-915 DEVIOT
DevNet	Cisco Certified DevNet Specialist–Webex	300-920 DEVWBX

You might notice that some of the specializations listed in Table 1-5 also appeared earlier in this chapter for the CCNP exams. In the original DevNet Professional model, those exams could count toward both the CCNP and DevNet Professional paths, and the DevNet Specialist exams could also stand on their own for specialist certification. Figure 1-5 shows the current Automation certification structure for comparison. As you can see, candidates progress through CCNA, CCNP, and CCIE Automation, while eligible professional-level exams also earn Specialist credentials.

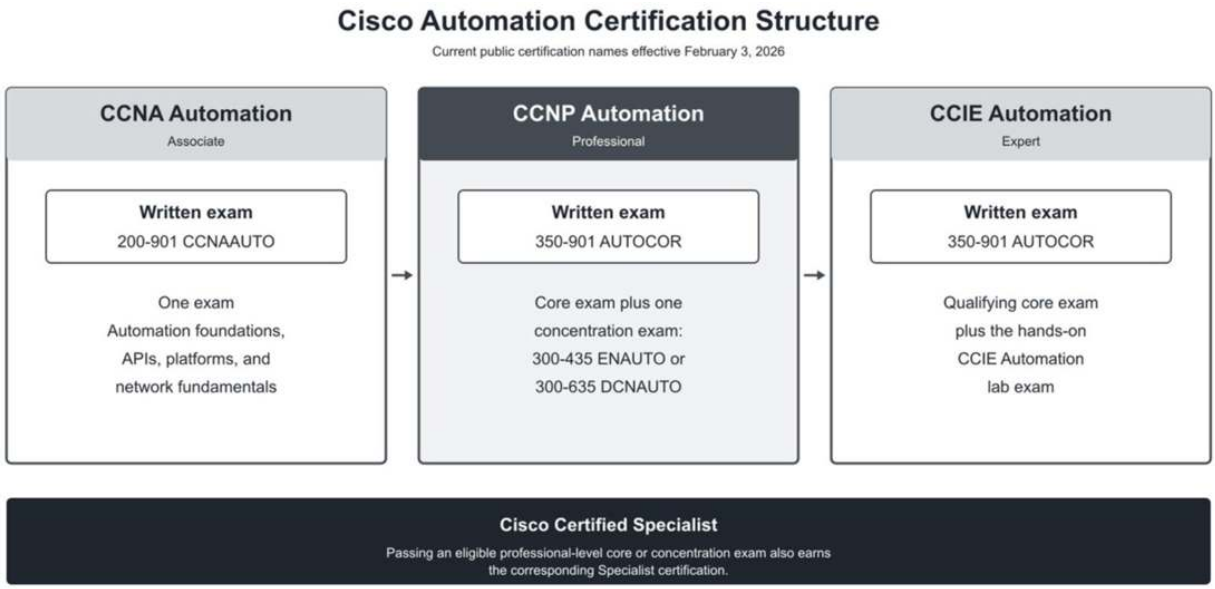


Figure 1-5 *Cisco Automation Certification Overview*

Cisco DevNet Overview

This section looks at the tools and resources available for Cisco Automation certification candidates. These tools help candidates to learn, practice, and share ideas and experiences.

The examples and tools discussed in this chapter are all available to use and practice at https://developer.cisco.com, which is the home for Cisco DevNet. This site provides a single place for network operators to go when looking to enhance or increase their skills with software development and automation on Cisco platforms.

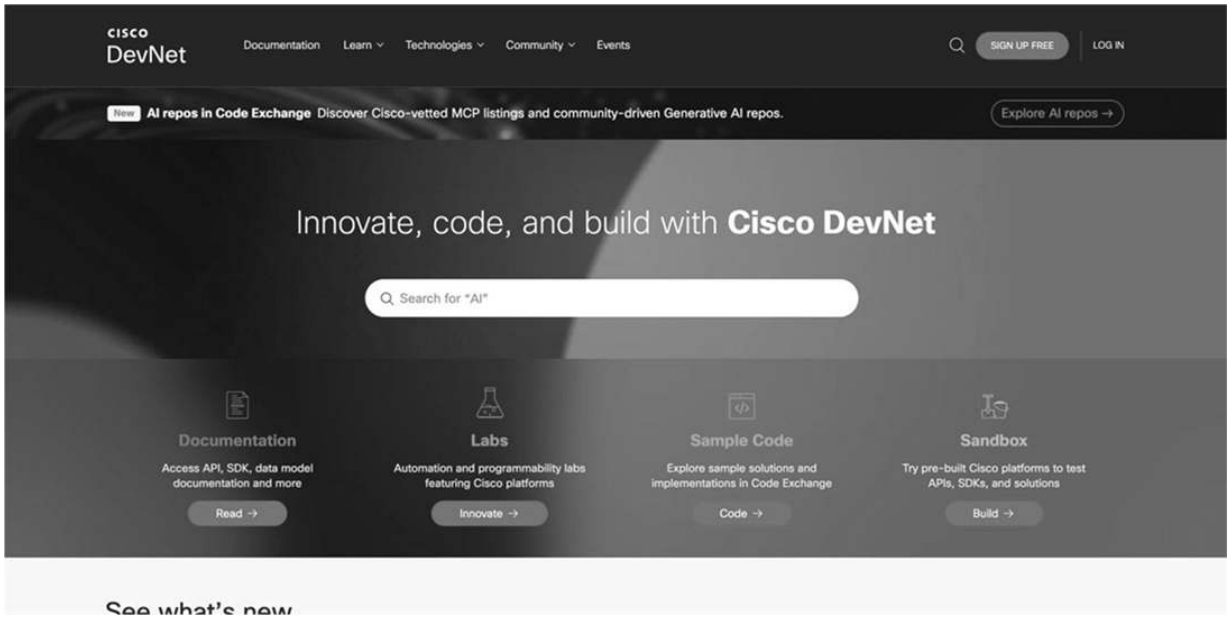


Figure 1-6 *Cisco DevNet Main Page, July 26, 2026*

The following sections highlight several kinds of DevNet resources that help candidates learn, practice, collaborate, and stay current:

- Learning
- Technologies
- Community

- Support Resources
- Events

The following sections discuss these resource areas individually.

Discover

Earlier versions of the DevNet site exposed a Discover page that highlighted different learning offerings. In practice, the useful idea here is the learning catalog itself: guided resources that walk you through technologies and associated API labs. Representative examples include Catalyst Center programmability, ACI programmability, Webex APIs, and introductory DevNet content.

When you choose a learning lab and start a module, DevNet tracks all your progress and allows you to go away and then come back and continue where you left off. This is an excellent feature if you are continuing your education over the course of multiple days or weeks. Being able to keep track of your progress means you can easily see what you have already learned and also determine what might be the next logical step in your learning journey.

Technologies

Earlier versions of the DevNet site also exposed a Technologies page that grouped labs and training by platform. The current portal organizes this material differently, but the practical takeaway is the same: you can still move directly from a Cisco technology area into the related developer resources, labs, and APIs.

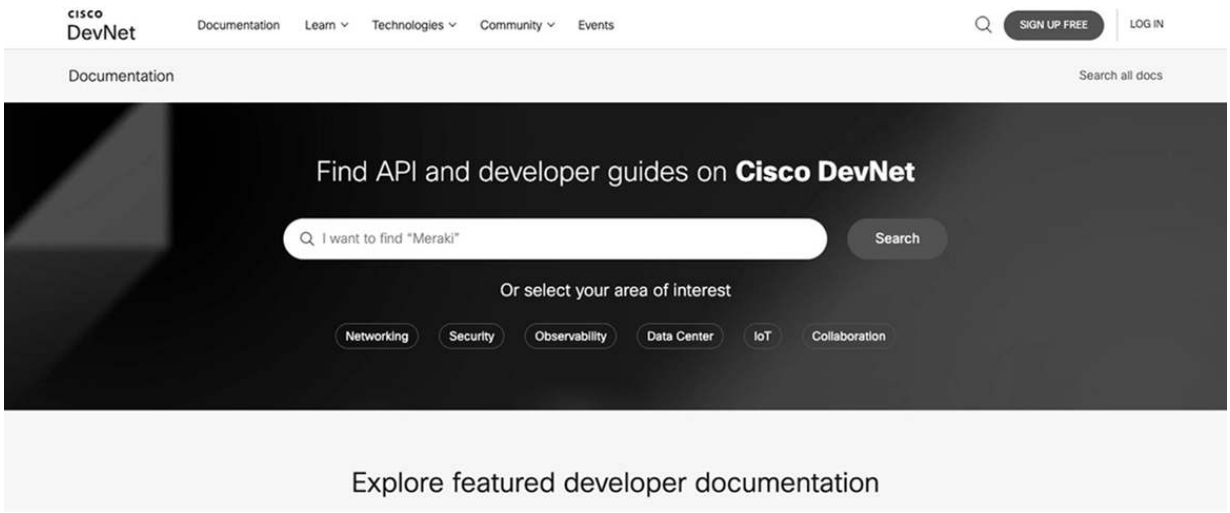


Figure 1-7 *DevNet Technologies Page*

> **Note**
>
> Available labs may differ from those shown in this chapter's figures. Please visit https://developer.cisco.com to see the latest content available and to interact with the current learning labs.

Community

Perhaps one of the most important sections of DevNet is the Community page, where you have access to many different people at various stages of learning. You can find DevNet ambassadors and evangelists to help at various stages of your learning journey. The Community page puts the latest events and news at your fingertips. This is also the place to read blogs, sign up for developer forums, and follow DevNet on all major social media platforms. This is the safe zone for asking any questions, regardless of how simple or complex they might seem. Everyone has to start somewhere. The DevNet Community page is the place

to start for all things Cisco and network programmability. Figure 1-8 shows some of the options currently available on the Community page.

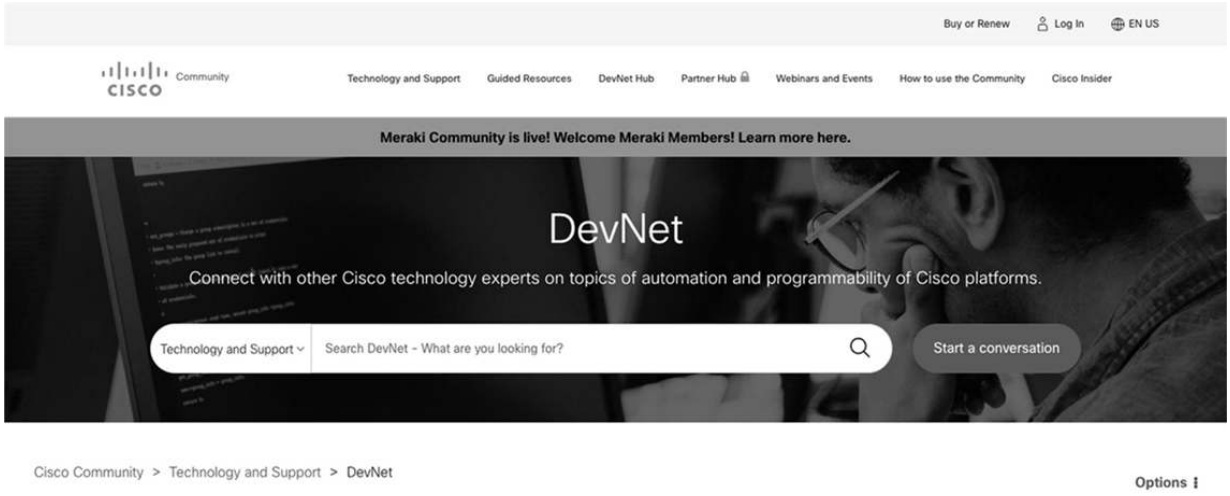


Figure 1-8 *DevNet Community Page*

Support

DevNet support is primarily community driven. For API questions and learning discussions, the developer community is the usual first stop, and it is often the fastest way to compare approaches with other developers working through similar examples. Product-specific operational issues still follow the normal Cisco support channels when a service contract applies, so it helps to separate learning questions from platform break-fix questions as you work.

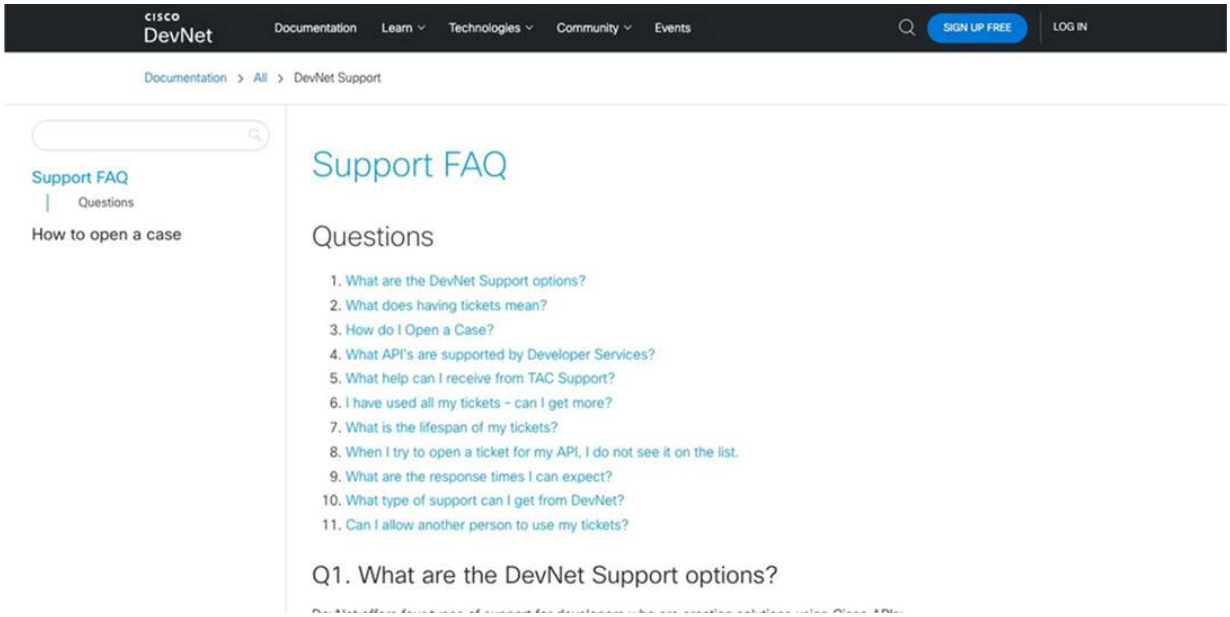


Figure 1-9 *DevNet Support FAQ*

Events

The Events page, shown in Figure 1-10, provides a list of current and past DevNet-related events. This is the place to look for upcoming developer programs, Cisco Live developer activities, and other DevNet appearances. Depending on the calendar, you may also see workshops, webinars, guided test drives, and community sessions that help turn self-study into hands-on practice.

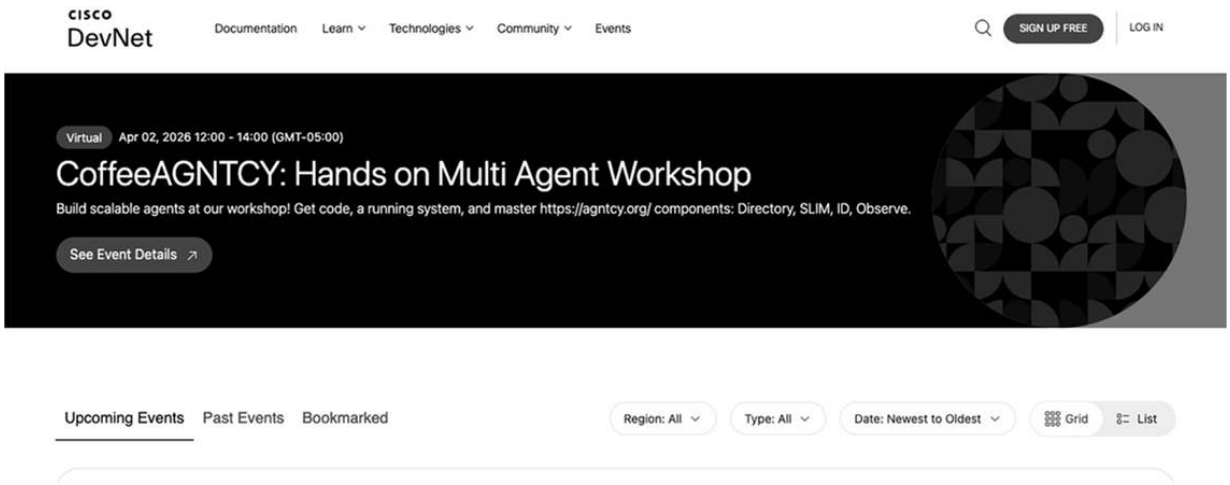


Figure 1-10 *DevNet Events Page*

> **Note**
>
> Keep in mind that the schedule shown in Figure 1-10 will differ from the schedule you see when you read this chapter.

DevNet gives customers the opportunity to learn modern development and engineering skills and also get hands-on experience in a safe environment. Earlier programs such as DevNet Express remain useful historical reference points, but Cisco now uses broader event and training labels across the DevNet ecosystem. The important idea has not changed: the point is to reduce the distance between reading about an API and actually trying it in a practical workflow.

DevNet Code Exchange

DevNet Code Exchange makes code available for consumption. Earlier DevNet materials often referred to Automation Exchange, but Cisco has since consolidated that idea under the Code Exchange banner. The spirit is the same: practical, use case-driven code examples that teams can study, adapt, and reuse to accelerate real work. The Walk, Run, and Fly framing used in this section comes from that earlier Automation Exchange material and remains useful conceptually even if the current Code Exchange organization differs. Think of it as a maturity model for the examples rather than a promise about the exact layout of the current website.

- Walk
- Run
- Fly

Figure 1-11 shows the landing page for DevNet Code Exchange. From there, you can browse shared projects, study implementation patterns, and contribute improvements back to the community once you have hardened a solution for wider use.

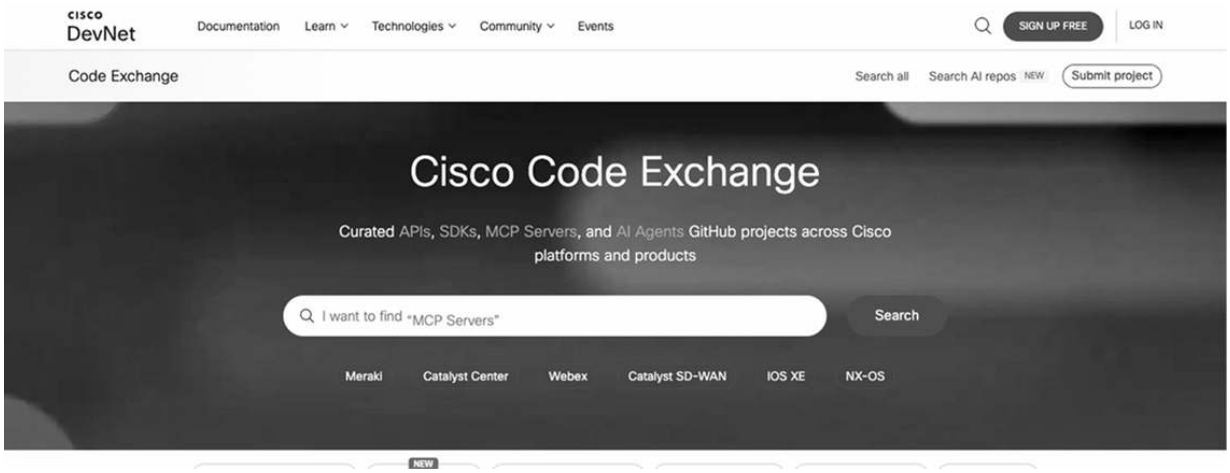


Figure 1-11 *DevNet Code Exchange*

When exploring the use case library, it is still helpful to think in terms of Walk, Run, and Fly maturity, along with automation lifecycle stage and network domain. In current Code Exchange views, the exact navigation may differ, but the conceptual grouping remains useful when you are deciding whether a project is read-only visibility, active configuration, or end-to-end operational automation.

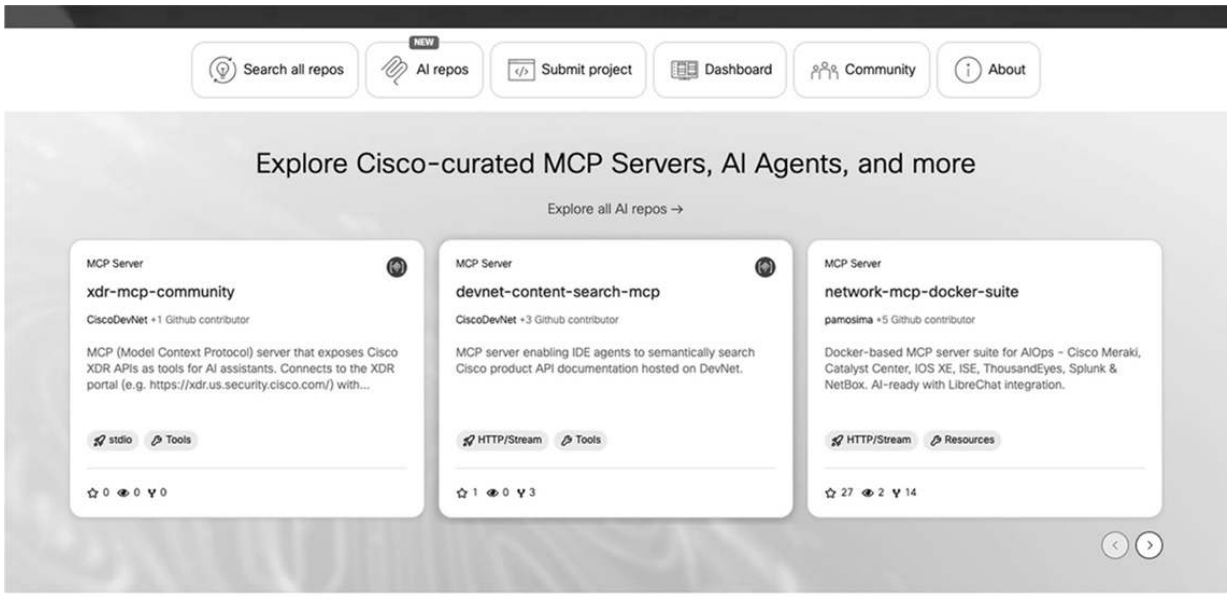


Figure 1-12 *DevNet Code Exchange Use Case Library*

The Walk category allows you to gain visibility and insights into your network. You can find various projects involving gathering telemetry and insight data in a read-only fashion. These projects can provide auditing capabilities to ensure the network's security and compliance. Because the projects are read-only, gathering the information has minimal risk of impacting a network negatively. You could, for example, use programmability to do a side-by-side configuration comparison to see what has changed in the configuration on a device. Using tools like this would be the next step past using the DevNet sandboxes to write code in a production environment.

The Run category in Code Exchange is where read/write actions start taking place in the network environment, such as when a network operations team begins to activate policies and express intent across different network domains. These projects often support self-service workflows while still helping teams stay aligned with security policies and operational standards.

Finally, the Fly category is for proactively managing applications, users, and devices by leveraging a DevOps workflow. With those projects, you can deploy applications using CI/CD pipelines while also

coordinating the network and the surrounding policy controls. It is a good example of where infrastructure automation stops being a side project and starts becoming part of normal engineering practice.

Table 1-6 *Automation Lifecycle*

Day	Function	Description
0	Install	Bringing devices into an initial operational state
1	Configure	Applying configurations to devices
2	Optimize	Implementing dynamic services, optimizing network behavior, and troubleshooting issues
N	Manage	Ensuring consistent and continuous operation of the network, with reduced risk and human error

Summary

This chapter provides a high-level overview of Cisco's career certifications and shows how candidates can build experience in the areas where they want to grow. It also refreshes the DevNet story with the current Automation transition, covers the specialist and professional paths that still matter for CCNA Automation candidates, and points to modern DevNet resources such as Code Exchange, Learning Labs, and the wider developer community.

Chapter 2. Software Development and Design

This chapter covers the following topics:

- **Software Development Lifecycle:** This section covers the Software Development Lifecycle (SDLC) and some of the most popular SDLC models, including Waterfall, Agile, and Lean.
- **Common Design Patterns:** This section covers common software design patterns, including the Model-View-Controller (MVC) and Observer design patterns.
- **Linux BASH:** This section covers key aspects of the Linux BASH shell and how to use it.
- **Software Version Control:** This section includes the use of version control systems in software development.
- **Git:** This section discusses the use of the Git version control system.
- **Conducting Code Review:** This section discusses using peer review to check the quality of software.

Are you a software developer? This has become an existential question for traditional network engineers, as programming and automation have become more pervasive in the industry. Professional programmers have picked up software integration with infrastructure gear as a new service or capability they can add to their applications. Traditional infrastructure engineers are being expected to know how to use APIs and automation tool sets to achieve more agility and speed in IT operations. The bottom line is that we are all being asked to pick up new skills to accomplish the business goals that keep us relevant and employed. This chapter discusses a few of the fundamental principles and tools that modern software development requires. You will learn about Agile, Lean, and common software design patterns that are used to enable a whole new operational model. In addition, you will see the importance of version control and how to use Git to collaborate with others and share your work with the world. These core concepts are essential to understanding the influence of software development methodologies as they pertain to infrastructure automation.

"Do I Know This Already?" Quiz

The "Do I Know This Already?" quiz allows you to assess whether you should read this entire chapter thoroughly or jump to the "Exam Preparation Tasks" section. If you are in doubt about your answers to these questions or your own assessment of your knowledge of the topics, read the entire chapter. Table 2-1 lists the major headings in this chapter and their corresponding "Do I Know This Already?" quiz questions. You can find the answers in Appendix A, "Answers to the 'Do I Know This Already?' Quiz Questions."

Table 2-1 *...'Do I Know This Already?' Section-to-Question Mapping*

Foundation Topics Section	Questions
Software Development Lifecycle	1, 2
Common Design Patterns	3, 4
Linux BASH	5, 6
Software Version Control	7
Git	8–10
Conducting Code Review	11

Caution

The goal of self-assessment is to gauge your mastery of the topics in this chapter. If you do not know the answer to a question or are only partially sure of the answer, you should mark that question as wrong for purposes of self-assessment. Giving yourself credit for an answer that you correctly guess skews your self-assessment results and might provide you with a false sense of security.

1. What is Waterfall?

   a. A description of how blame flows from management on failed software projects

   b. A type of SDLC

   c. A serial approach to software development that relies on a fixed scope

   d. All of the above

2. What is Agile?

   a. A form of project management for Lean

   b. An implementation of Lean for software development

   c. A strategy for passing the CCNAAUTO exam

   d. A key benefit of automation in infrastructure

3. The Model-View-Controller pattern is often used in which of the following applications? (Choose three.)

   a. Web applications with graphical interfaces

   b. Client/server applications with multiple client types

   c. PowerShell scripts

   d. Django

4. Which of the following are true of the Observer pattern? (Choose two.)

   a. It is a publisher/subscriber pattern.

   b. It is a multicast pattern.

   c. It is used for configuration management applications and event handling.

d. It is not commonly used in infrastructure application design.

5. What does BASH stand for?

a. Born Again Shell

b. Basic Shell

c. Bourne Again Shell

d. None of the above

6. Which of the following is the best option for displaying your current environment variables?

a. env | cat >env.txt

b. env | more

c. export $ENV | cat

d. echo env

7. Which of the following is true of software version control?

a. It is a naming convention for software releases.

b. It is also known as source code management.

c. It is the same thing as BitKeeper.

d. None of the above are true.

8. Who created Git?

a. Junio Hamano

b. Marc Andreesen

c. John Chambers

d. Linus Torvalds

9. What are the three main structures tracked by Git?

a. Index, head, and local repo

b. Local workspace, index, and local repository

c. Remote repository, head, and local index

d. None of the above

10. What command do you use to add the specific filename file.py to the Git index?

a. git add .

b. git index file.py

c. git index add .

d. git add file.py

11. Which is one of the key benefits of conducting a code review?

a. It helps you create higher-quality software

b. You can find weak programmers who need more training

c. It can be used to identify defects in software that are obvious

d. None of the above

Foundation Topics

Software Development Lifecycle

Anyone can program. Once you learn a programming language's syntax, it's just a matter of slapping it all together to make your application do what you want it to do, right? The reality is, software needs to be built using a structure to give it sustainability, manageability, and coherency. You may have heard the phrase "cowboy coding" to refer to an unstructured software project, where there is little formal design work, and the programmer just sort of "shoots from the hip" and slaps code in with little or no forethought. This is a path that leads straight to late-night support calls and constant bug scrubbing. Heaven forbid if you inherit a ball of spaghetti like this and you are asked to try to fix, extend, or modernize it. You will more than likely be updating your resume or packing your parachute for a quick escape.

To prevent problems from slapdash approaches such as cowboy coding, disciplines such as architecture and construction establish rules and standards that govern the process of building. In the world of software, the Software Development Lifecycle (SDLC) provides sanity by providing guidance on building sustainable software packages. SDLC lays out a plan for building, fixing, replacing, and making alterations to software.

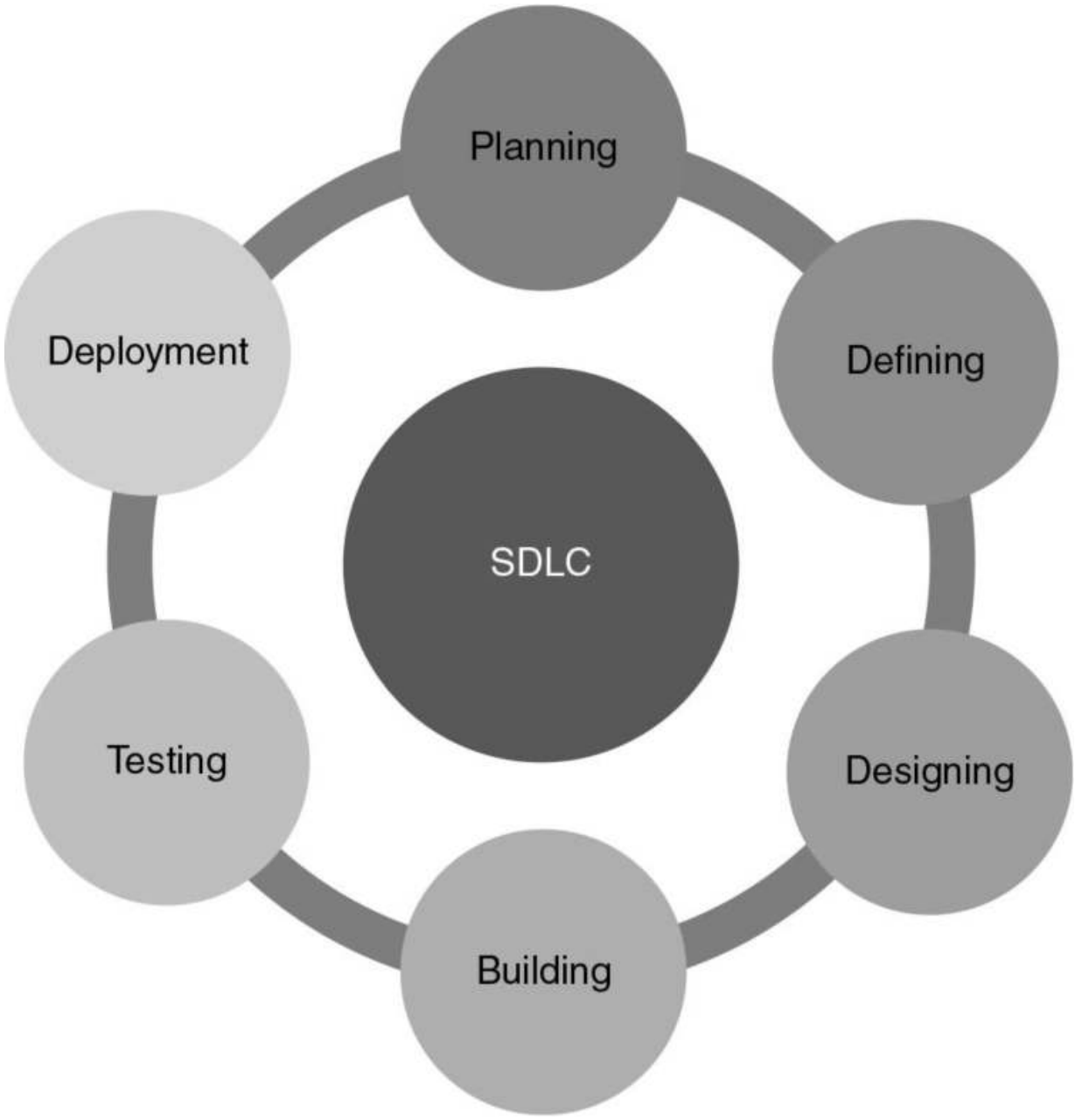


Figure 2-1 *...Software Development Lifecycle*

As shown in Figure 2-1, these are the stages of the SDLC:

- **Stage 1—Planning:** Identify the current use case or problem the software is intended to solve. Get input from stakeholders, end users, and experts to determine what success looks like. This stage is also known as *requirements analysis*.
- **Stage 2—Defining:** This stage involves analyzing the functional specifications of the software—basically defining what the software is supposed to do.
- **Stage 3—Designing:** In this phase, you turn the software specifications into a design specification. This is a critical stage as stakeholders need to be in agreement in order to build the software appropriately; if they aren't, users won't be happy, and the project will not be successful.
- **Stage 4—Building:** Once the software design specification is complete, the programmers get to work on making it a reality. If the previous stages are completed successfully, this stage is often considered the easy part.
- **Stage 5—Testing:** Does the software work as expected? In this stage, the programmers check for bugs and defects. The software is continually examined and tested until it successfully meets the

original software specifications.

- **Stage 6—Deployment:** During this stage, the software is put into production for the end users to put it through its paces. Deployment is often initially done in a limited way to do any final tweaking or detect any missed bugs. Once the user has accepted the software and it is in full production, this stage morphs into maintenance, where bug fixes and software tweaks or smaller changes are made at the request of the business user.

> **Note**
>
> ISO/IEC 12207 is the international standard for software lifecycle processes, and there are numerous organizations around the globe that use it for certification of their software development efforts. It is compatible with any SDLC models and augments them from a quality and process assurance standpoint. It does not, however, replace your chosen SDLC model.

There are quite a few SDLC models that further refine the generic process just described. They all use the same core concepts but vary in terms of implementation and utility for different projects and teams. The following are some of the most popular SDLC models:

- Waterfall
- Lean
- Agile
- Iterative model
- Spiral model
- V model
- Big Bang model
- Prototyping models

Luckily, you don't need to know all of these for the 200-901 CCNAAUTO exam. The following sections cover the ones you should know most about: Waterfall, Lean, and Agile.

Waterfall

Back in the 1950s, when large companies started to purchase large mainframe computers to crunch numbers, no one really knew how to run an IT organization. It really wasn't anything that had been done before, and for the most part, computers were only really understood by an elite group of scientists. Programming a mainframe required structure and a process. This caused a problem for businesses looking to tap into the capabilities of these new systems since there wasn't a well-known method to create business applications. So they looked around at other industries for guidance.

The construction industry was booming at the time. The construction industry followed a rigid process in which every step along the way was dependent on the completion of the previous step in the process. If you want to end up with a building that stays standing and meets the original design, you can't start construction until you have a plan and analyze the requirements for the building. This thought process mapped nicely to software development, and the complexity of designing and constructing a building was similar to that of creating software applications. Waterfall, which is based on the construction industry approach, became one of the most popular SDLC approaches.

As illustrated in Figure 2-2, Waterfall is a serial approach to software development that is divided into phases:

- **Requirements/analysis:** Software features and functionality needs are cataloged and assessed to determine the necessary capabilities of the software.
- **Design:** The software architecture is defined and documented.
- **Coding:** Software coding begins, based on the previously determined design.
- **Testing:** The completed code is tested for quality and customer acceptance.
- **Maintenance:** Bug fixes and patches are applied.

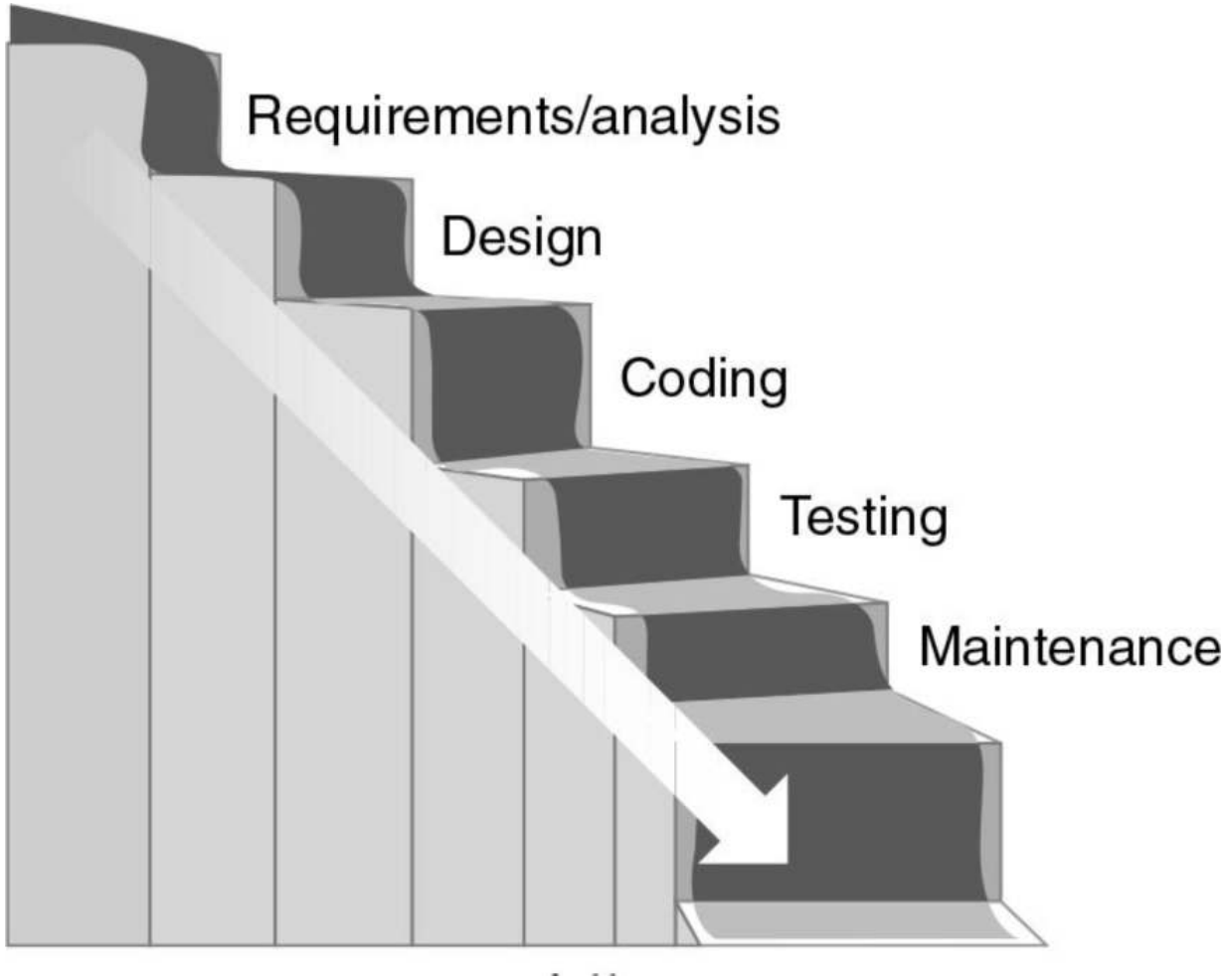


Figure 2-2 …*Waterfall*

While this approach has worked successfully over the years, a number of shortcomings have become weaknesses in this approach. First, the serial nature of Waterfall, while easy to understand, means that the scope of a software project is fixed at the design phase. In construction, making changes to the first floor of a building after you have begun the fifth floor is extremely difficult—and may even be impossible unless you knock down the building and start from scratch. In essence, the Waterfall approach does not handle change well at all. When you finally get to the coding phase of the application development process, you might learn that the feature you are building isn't needed anymore or discover a new way of accomplishing a design goal; however, you cannot deviate from the predetermined architecture without redoing the analysis and design. Unfortunately, it is often more painful to start over than to keep building. It is similar to being stuck building a bridge over a river that no one needs anymore.

The second aspect of Waterfall that is challenging is that value is not achieved until the end of the whole process. We write software to automate some business function or capability—and value is only realized when the software is in production and producing results. With the Waterfall approach, even if you are

halfway done with a project, you still have no usable code or value to show to the business. Figure 2-3 shows this concept.

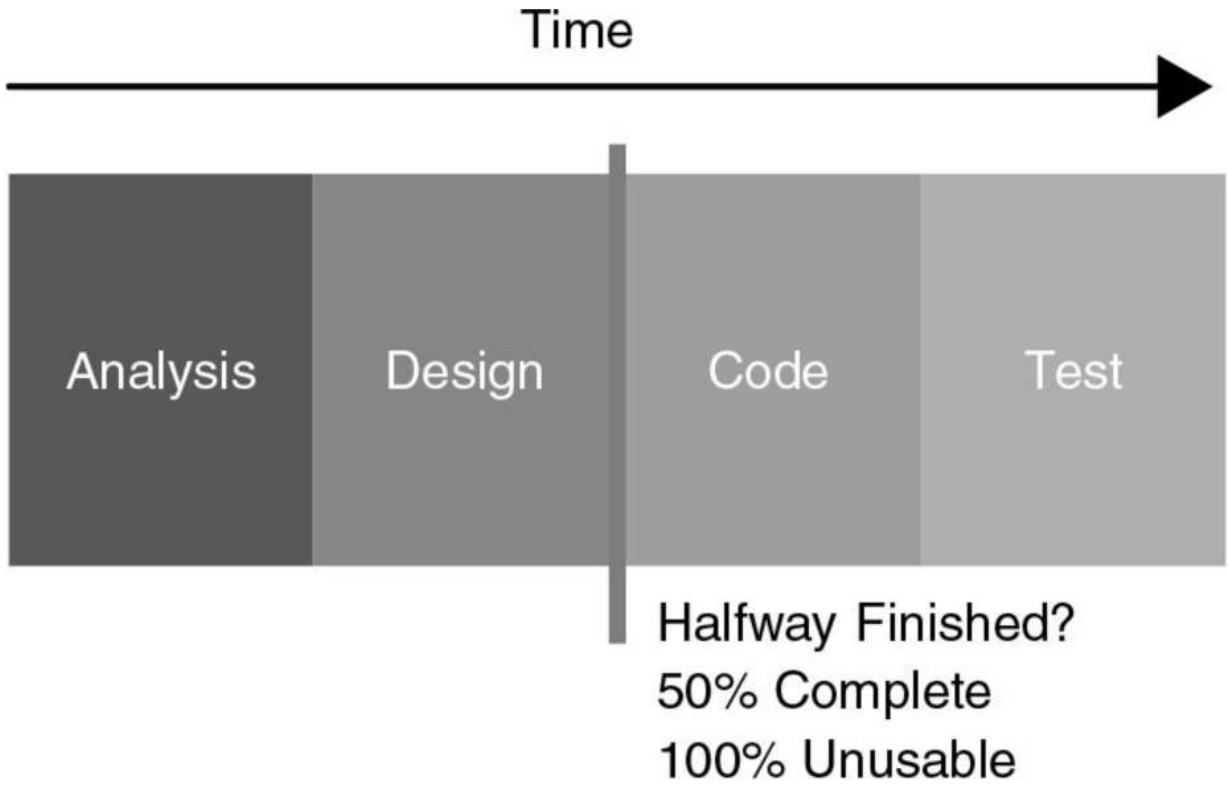


Figure 2-3 *...The Value Problem of Waterfall*

The third aspect of Waterfall that is challenging is quality. As mentioned earlier, time is the enemy when it comes to delivering value. If we had unlimited time, we could create perfect software every time, but we simply don't live in that world. When software developers run out of time, testing often suffers or is sacrificed in the name of getting the project out the door.

The three challenges for Waterfall led to the development of a new way of creating software that was faster, better, and more adaptive to a rapidly changing environment.

Lean

After World War II, Japan was in desperate need of rebuilding. Most of Japan's production capabilities had been destroyed, including those in the auto industry. When Japan tackled this rebuilding, it didn't concentrate on only the buildings and infrastructure; it looked at ways to do things differently. Out of this effort, the Toyota Production System (TPS) was born. Created by Taiichi Ohno and Sakichi Toyoda (founder of Toyota), this management and manufacturing process focuses on the following important concepts:

- **Elimination of waste:** If something doesn't add value to the final product, get rid of it. There is no room for wasted work.
- **Just-in-time:** Don't build something until the customer is ready to buy it. Excess inventory wastes resources.
- **Continuous improvement (Kizan):** Always improve your processes with lessons learned and communication.

While these concepts seem glaringly obvious and practical, TPS was the first implementation of these principles as a management philosophy. TPS was the start of the more generalized Lean manufacturing approach that was introduced to the Western world in 1991 through a book written by Womack, Jones, and Roos, *The Machine That Changed the World*. This book was based on a five-year study MIT conducted on TPS, and it has been credited with bringing Lean concepts and processes beyond the auto industry.

Why spend this time talking about moldy old management books? Lean led to Agile software development, which has served as a lightning rod of change for IT operations.

Agile

Agile is an application of Lean principles to software development. With Agile, all the lessons learned in optimizing manufacturing processes have been applied to the discipline of creating software. In 2001, 17 software developers converged on the Snowbird resort in Utah to discuss new lightweight development methods. Tired of missing deadlines, endless documentation, and the inflexibility of existing software development practices, these Agile pioneers created the "Manifesto for Agile Software Development," which codifies the guiding principles for Agile development practices. The following 12 principles are the core of the Agile Manifesto:

- Customer satisfaction is provided through early and continuous delivery of valuable software.
- Changing requirements, even in late development, are welcome.
- Working software is delivered frequently (in weeks rather than months).
- The process depends on close, daily cooperation between business stakeholders and developers.
- Projects are built around motivated individuals, who should be trusted.
- Face-to-face conversation is the best form of communication (co-location).
- Working software is the principal measure of progress.
- Sustainable development requires being able to maintain a constant pace.
- Continuous attention is paid to technical excellence and good design.
- Simplicity—the art of maximizing the amount of work not done—is essential.
- The best architectures, requirements, and designs emerge from self-organizing teams.
- A team regularly reflects on how to become more effective and adjusts accordingly.

These core tenets were the main spark of the Agile movement. Mary Poppendieck and Tom Poppendieck wrote *Lean Software Development: An Agile Toolkit* in 2003, based on the principles of the Agile Manifesto and their many years of experience developing software. This book is still considered one of the best on the practical uses of Agile.

Developing software through Agile results in very different output than the slow serial manner used with Waterfall. With Waterfall, a project is not "finished" and deployable until the very end. With Agile, the time frame is changed: Agile uses smaller time increments (often 2 weeks), or "sprints," that encompass the full process of analysis, design, code, and test but on a much smaller aspect of an application. The goal is to finish a feature or capability for each sprint, resulting in a potentially shippable incremental piece of software. Therefore, with Agile, if you are 40% finished with a project, you have 100% usable code. Figure 2-4 shows how this process looks on a timeline.

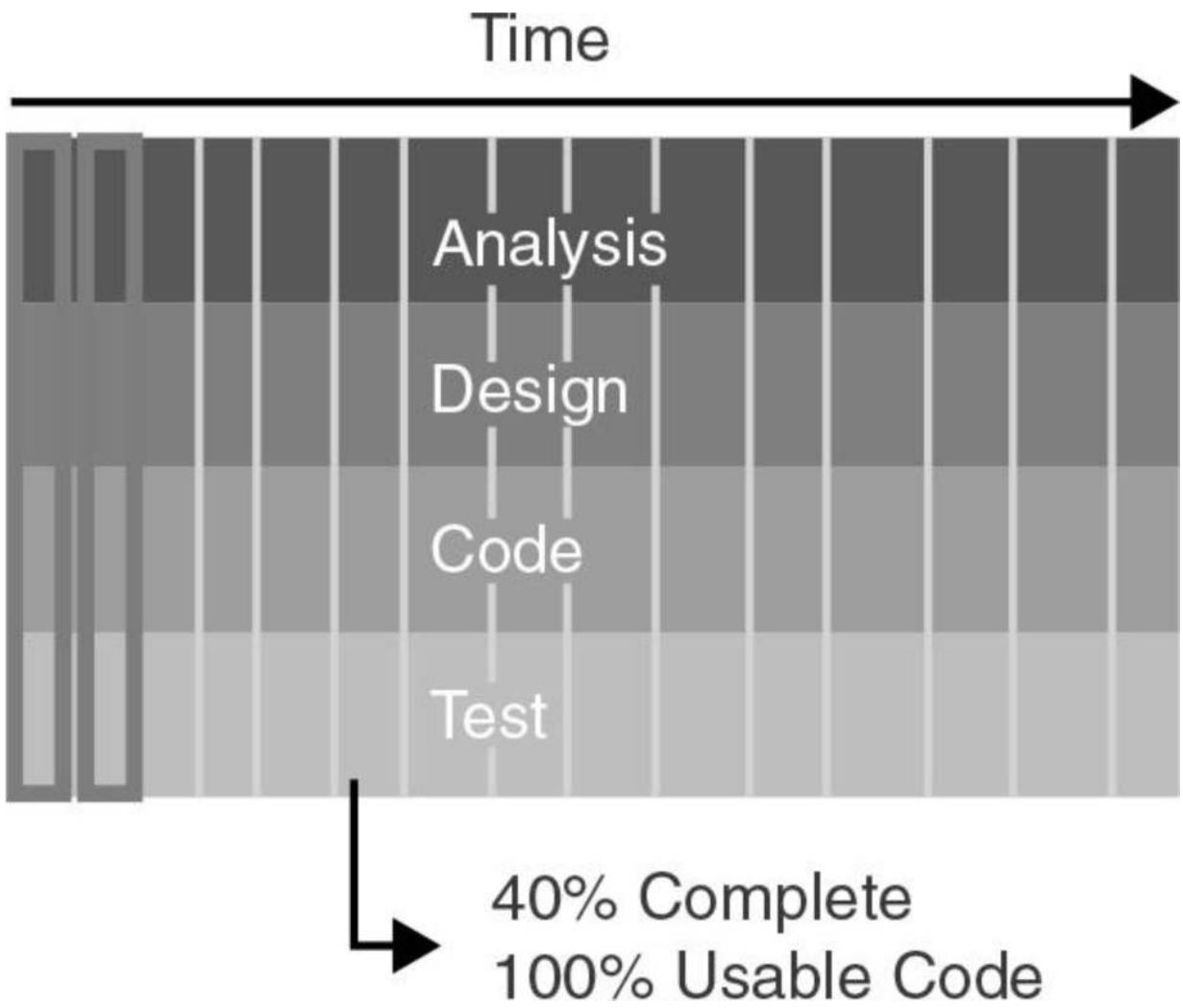


Figure 2-4 *...Agile Development Practices*

By leveraging Agile, you can keep adding value immediately and nimbly adapt to change. If a new capability is needed in the software, or if a feature that was planned is determined to no longer be necessary, the project can pivot quickly and make those adjustments.

Common Design Patterns

When creating software, you will often run into the same problem over and over again. You don't want to reinvent the wheel each time you need a rolling thing to make something move. In software engineering, many common design paradigms have already been created, and you can reuse them in your software project. These design patterns make you faster and provide tried-and-true solutions that have been tested and refined. The following sections introduce a couple of design patterns that are really useful for network automation projects: the Model-View-Controller (MVC) and Observer patterns. While there are many more that you may be interested in learning about, these are the ones you will most likely see on the 200-901 CCNAAUTO exam.

Model-View-Controller (MVC) Pattern

The Model-View-Controller (MVC) pattern was one of the first design patterns to leverage the separation of concerns (SoC) principle. The SoC principle is used to decouple an application's interdependencies and

functions from its other parts. The goal is to make the various layers of the application—such as data access, business logic, and presentation (to the end user)—modular. This modularity makes the application easier to construct and maintain while also allowing the flexibility to make changes or additions to business logic. It also provides a natural organization structure for a program that anyone can follow for collaborative development. If you have used a web-based application, more than likely the app was constructed using an MVC pattern.

> **Note**
>
> Numerous web frameworks use MVC and MVC-derived concepts across many programming languages. Django (Python) is a classic MVC implementation (technically MVT). Ruby on Rails (Ruby), Laravel (PHP), and ASP.NET MVC (C#) all use the pattern directly. Many modern JavaScript frameworks such as Angular, React, and Vue use the broader separation-of-concerns idea but with their own variations (component + service patterns). Flask (Python) is a microframework that gives you the building blocks to implement MVC yourself.

The classical MVC pattern has three main parts:

- **Model:** The model is responsible for retrieving and manipulating data. It is often tied to some type of database but could be data from a simple file. It conducts all data operations, such as select, insert, update, and delete operations. The model receives instructions from the controller.
- **View:** The view is what the end users see on the devices they are using to interact with the program. It could be a web page or text from the command line. The power of the view is that it can be tailored to any device and any representation without changing any of the business logic of the model. The view communicates with the controller by sending data or receiving output from the model through the controller. The view's primary function is to render data.
- **Controller:** The controller is the intermediary between what the user sees and the backend logic that manipulates the data. The role of the controller is to receive requests from the user via the view and pass those requests on to the model and its underlying data store.

Figure 2-5 shows the interactions between components of the MVC pattern.

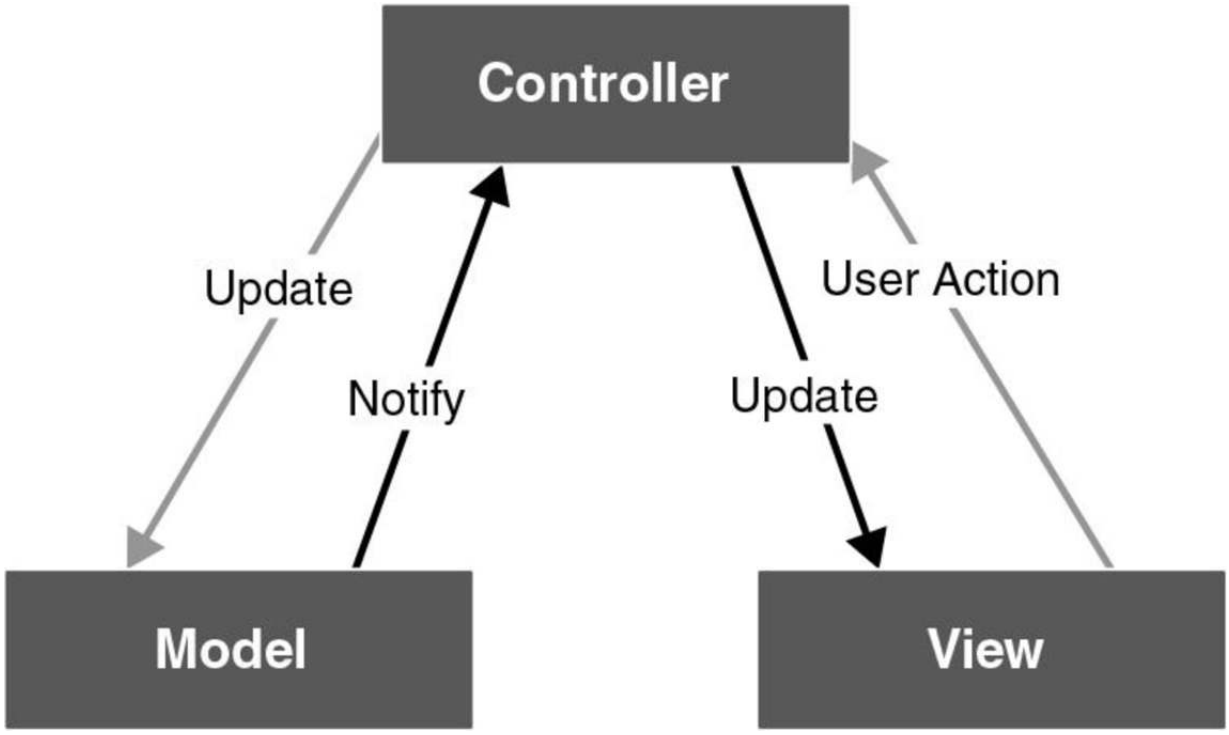


Figure 2-5 *...MVC Pattern Interactions*

Observer Pattern

The Observer pattern was created to address the problem of sharing information between one object to many other objects. This type of pattern describes a very useful behavior for distributed systems that need to share configuration information or details on changes as they happen. The Observer pattern is actually very simple and consists of only two logical components (see Figure 2-6):

- **Subject:** The subject refers to the object state being observed—in other words, the data that is to be synchronized. The subject has a registration process that allows other components of an application or even remote systems to subscribe to the process. Once registered, a subscriber is sent an update notification whenever there is a change in the subject's data so that the remote systems can synchronize.

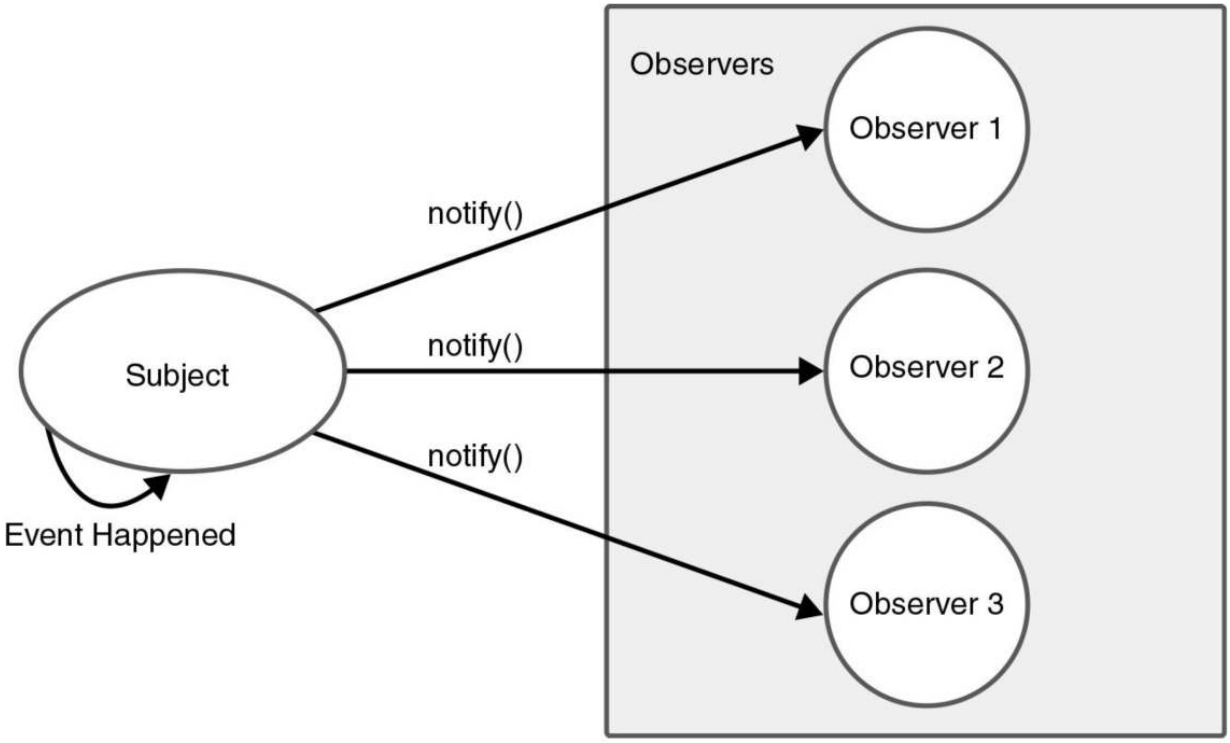


Figure 2-6 *...Observer Pattern*

- **Observer:** The observer is the component that registers with the subject to allow the subject to be aware of the observer and how to communicate to it. The only function of the observer is to synchronize its data with the subject when called. The key thing to understand about the observer is that it does not use a polling process, which can be very inefficient with a larger number of observers registered to a subject. Updates are push only.

The Observer pattern is often used to handle communications between the model and the view in the MVC pattern. Say, for example, that you have two different views available to an end user. One view provides a bar graph, and the other provides a scatter plot. Both use the same data source from the model. When that data changes or is updated, the two views need to be updated. This is a perfect job for the Observer pattern.

Linux BASH

Knowing how to use Linux BASH is a necessary skill for working with open-source technologies as well as many of the tools you need to be proficient with to be successful in the development world. Linux has taken over the development world, and even Microsoft has jumped into the game by providing the Windows Subsystem for Linux for Windows 11. For the CCNAAUTO exam, you need to know how to use BASH and be familiar with some of the key commands.

Getting to Know BASH

BASH is a shell, and a shell is simply a layer between a user and the internal workings of an operating system. A user can use the shell to input commands that the operating system will interpret and perform. Before graphical user interfaces (GUI) became common, the shell reigned supreme, and those who knew its intricacies were revered as some tech wizards. Today that skill is still in high demand, and without it you will find yourself struggling as the GUI simply doesn't make possible many of the powerful operations available through the shell.

While there are many shells you can use, BASH, which stands for Bourne Again Shell, is one of the most popular. It has been around since 1989 and is the default shell on most Linux operating systems. Until recently, it was also the default for the Mac operating system, but Apple has replaced BASH with Z shell. The commands and syntax you will learn with BASH are transferable to Z shell, as it was built to maintain compatibility with BASH.

BASH is not only a shell for command processing using standard Linux operating system commands. It can also read and interpret scripts for automation. These capabilities are beyond the scope of what you need to know for the CCNAAUTO exam but would be worth looking into as you continue your journey as these automation scripts are where BASH really shines. Like all other UNIX shells, BASH supports features such as piping (which involves feeding output from one command as the input of another command), variables, evaluation of conditions, and iteration (repeated processing of a command with **if** statements). You also have a command history as part of the shell, where you can use the arrow keys to cycle through and edit previous commands.

UNIX platforms such as Linux and macOS have built-in documentation for each command the operating system uses. To access help for any command, type **man** (short for manual) and then the command you are curious about. The output gives you a synopsis of the command, any optional flags, and required attributes. Example 2-1 shows the man page for the **man** command.

Example 2-1 *...Example of the Manual Page for the **man** Command*

```
2$ man man

man(1)                                                          man(1)

NAME

     man - format and display the on-line manual pages

SYNOPSIS
     man  [-acdfFhkKtwW]  [--path]  [-m system] [-p string] [-C config_file]
     [-M pathlist] [-P pager] [-B browser] [-H htmlpager] [-S  section_list]
     [section] name ...

DESCRIPTION
     man formats and displays the on-line manual pages.  If you specify sec-
     tion, man only looks in that section of the manual.  name  is  normally
     the  name of the manual page, which is typically the name of a command,
     function, or file.  However, if name contains  a  slash  (/)  then  man
     interprets  it  as a file specification, so that you can do man ./foo.5
     or even man /cd/foo/bar.1.gz.

     See below for a description of where man  looks  for  the  manual  page
     files.

<output cut for brevity>
```

Not every command is intended to be run with user-level privileges. You can temporarily upgrade your privileges by prepending the **sudo** command before you execute a command that needs higher-level access. You will often be prompted for a password to verify that you have the right to use **sudo**. You need to be careful when using **sudo**, as the whole idea of reduced privileges is to increase security and prevent average users from running commands that are dangerous. Use **sudo** only when required, such as when you need to kick off an update for your Linux distribution, which you do by using the **apt-get update** command:

```
$ sudo apt-get update
```

As mentioned earlier, one of the most powerful features of BASH is something called piping. This feature allows you to string together commands. For example, the **cat** command displays the contents of a file to the screen. What if a file contains too much to fit on a single screen? The **cat** command will happily spew every character of the file at the screen until it reaches the end, regardless of whether you could keep up with it. To address this, you can pipe the output of **cat** to the **more** command to stream the content from **cat** to **more**, which gives you a prompt to continue one page at a time. To use the piping functionality, you use the pipe character (|) between commands, as shown in Example 2-2.

Example 2-2 *...Output of the **cat** Command Piped to the **more** Command*

```
$cat weather.py | more

import json
import urllib.request
from pprint import pprint


def get_local_weather():

   weather_base_url =
'https://forecast.weather.gov/MapClick.php?FcstType=json&'

   places = {
      'Austin': ['30.3074624', '-98.0335911'],
      'Portland': ['45.542094', '-122.9346037'],
      'NYC': ['40.7053111', '-74.258188']
   }

   for place in places:
      latitude, longitude = places[place][0], places[place][1]
      weather_url = weather_base_url + "lat=" + latitude + "&lon=" + longitude
      # Show the URL we use to get the weather data. (Paste this URL into your
      browser!)
      # print("Getting the current weather for", place, "at", weather_url, ":")

      page_response = urllib.request.urlopen(weather_url).read()
<output cut for brevity>
```

Directory Navigation

A UNIX-based file system has a directory tree structure. The top of the tree is called the *root* (as it's an upside-down tree), and you use the forward slash (/) to refer to root. From root you have numerous directories, and under each directory you can have more directories or files. Figure 2-7 shows a UNIX directory structure.

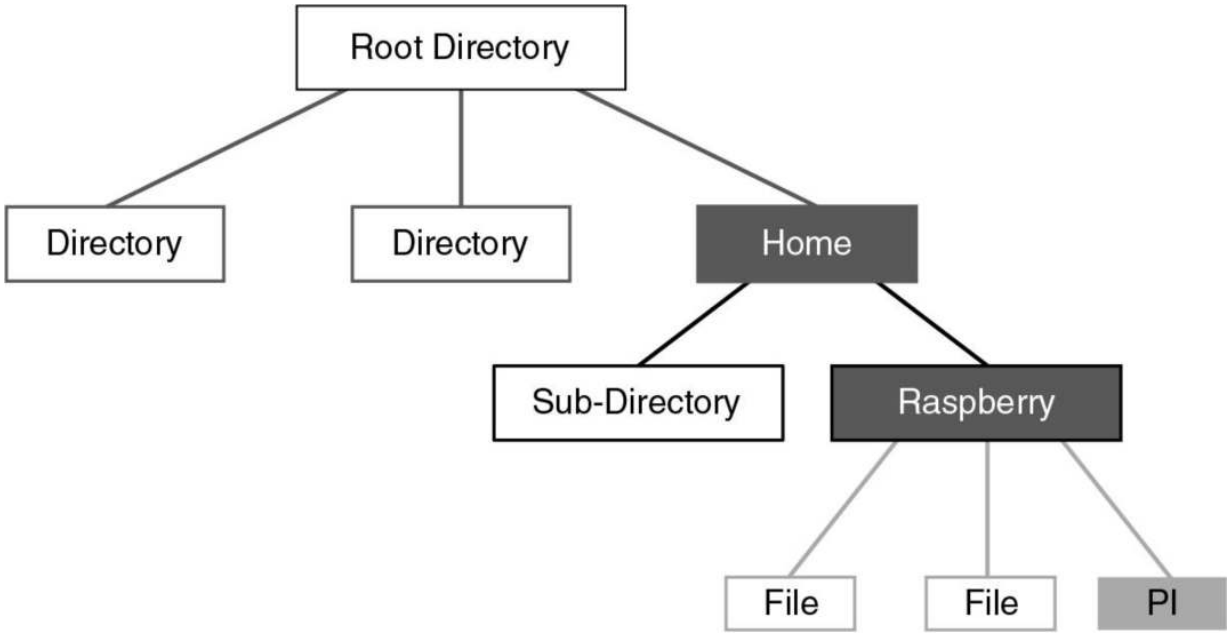


Figure 2-7 *…UNIX Directory Structure*

Whenever you call a file, you have to supply its path. Everything you execute in UNIX is in relationship to root. To execute a file in the directory you are in, you can use **./filename.sh**, where the leading **.** is simply an alias for the current directory.

In addition to the root file system, each user has a home directory that the user controls and that stores the user's individual files and applications. The full path to the home directory often looks something like /home/*username* on Linux and /Users/*username* on macOS, but you can also use the tilde shortcut (~/) to reference the home directory.

The following sections describe some of the commands most commonly used to interact with the BASH shell and provide examples of their options and use.

cd

The **cd** command is used to change directories and move around the file system. You can use it as follows:

$ cd /	Changes directory to the root directory
$ cd /home/username	Changes directory to the /home/username directory
$ cd test	Changes directory to the test folder
$ cd ..	Moves up one directory

pwd

If you ever get lost while navigating around the file system, you can use the **pwd** command to print out your current working directory path. You can use it as follows:

$ pwd	Print your current working directory

ls

Once you have navigated to a directory, you probably want to know what is in it. The **ls** command gives you a list of the current directory. If you execute it without any parameters, it just displays whatever is in the directory. It doesn't show any hidden files (such as configuration files). Anything that starts with a **.** does not show up in a standard directory listing, and you need to use the **-a** flag to see all files, including

hidden files. By using the **-l** flag, you can see permissions and the user and group that own the file or directory. You can also use the wildcard ***** to list specific filename values; for example, to find any files with test as a part of the name, you can use **ls *test***, which would match both 1test and test1. You can use the **ls** command as follows:

$ ls	Lists files and directories in the current working directory
$ ls -a	Lists everything in the current directory, including hidden files
$ ls /home/username	Lists everything in the /home/username directory
$ ls -l	Lists permissions and user and group ownership
$ ls -F	Displays files and directories and denotes which are which

mkdir

To create a directory, you use the **mkdir** command. If you are in your home directory or in another directory where you have the appropriate permissions, you can use this command without **sudo**. You can use the **mkdir** command as follows:

$ mkdir test	Makes a new directory called test in the current working directory if you have permission
$ mkdir /home/username/test	Makes a new directory called test at /home/username/test

File Management

Working with files is easy with BASH. There are just a few commands that you will use often, and they are described in the following sections.

cp

The purpose of the **cp** command is to copy a file or folder someplace. It does not delete the source file but instead makes an identical duplicate. When editing configuration files or making changes that you may want to roll back, you can use the **cp** command to create a copy as a sort of backup. The command requires several parameters: the name of the file you want to copy and where you want to copy it to and the name of the copy. When copying the contents of a folder, you need to use the **-r**, or recursive, flag. You can use the **cp** command as follows:

$ cp sydney.txt sydney2.txt	Copies a file called sydney.txt from the current directory and names the copy sydney2.txt
$ cp /home/username/	Copies a file as described above but using the full path and
sydney.txt ~/sydney2.txt	the home directory path
$ cp -r folder folder.old	Copies a folder

mv

The **mv** command allows you to move a file or folder from one directory to another, and it is also used to rename files or folders from the command line, as BASH does not have a dedicated renaming function. The **mv** command takes a source and destination, just as **cp** does. You can use the **-i** flag to create an interactive option prompt when moving files that exist at the destination. The **-f** flag forces the move and overwrites any files at the destination. Wildcards also work to select multiple source files or directories. You can use the **mv** command as follows:

$ mv caleb.txt calebfinal.txt	Renames a file called caleb.txt to calebfinal.txt
$ mv /home/username/ caleb.txt ~/calebfinal.txt	Renames a file as described above but using full paths
$ mv -i * /home/ username/new/	Moves all files and directories in the current folder to a directory called new

rm

To delete a file or directory, you use the **rm** command. If the item you are deleting is a file or an empty directory, you just need to supply the name and press Enter. On the other hand, if you try to delete a directory that has files in it, **rm** tells you that the directory is not empty. In that case, you can use the **-rf** flag to force the deletion. You can use the **rm** command as follows:

$ rm test.txt	Deletes the file test.txt in the current working directory
$ rm -rf test	Forces the deletion of the folder test and everything in it

touch

The **touch** command is used to create a file and/or change the timestamps on a file's access without opening it. This command is often used when a developer wants to create a file but doesn't want to put any content in it. You can use the **touch** command as follows:

$ touch emptyfile.txt	Creates an empty file named emptyfile.txt
$ touch file{1..20}.txt	Bulk creates files from file1.txt to file20.txt

cat

The **cat** (which stands for *concatenate*) command allows you to view or create files and also pipe to other commands. It's one of the most useful commands in UNIX when it comes to working with files. You can use the **cat** command as follows:

$cat file1.txt	Displays the contents of file1.txt
$cat file1.txt | more	Displays the contents of file1.txt and pipes the output to **more** to add page breaks
$cat >file2.txt	Sends a user's typed or copied content from the command line to file2.txt

Environment Variables

BASH environment variables contain information about the current session. Environment variables are available in all operating systems and are typically set when you open your terminal from a configuration file associated with your login. You set these variables with similar syntax to how you set them when programming. You do not often use these variables directly, but the programs and applications you launch do. You can view all of your currently set environment variables by entering the **env** command. Since there can be more entries than you have room to display on a single terminal page, you can pipe the results to the **more** command to pause between pages:

$env | more	Shows all environment variables with page breaks

If you execute this command, you are likely to notice a lot of keywords with the = sign tied to values. One environment variable that you use every time you execute a command is the PATH variable. This is where

your shell looks for executable files. If you add a new command and can't execute it, more than likely the place where the command was copied is not listed in your PATH. To view any variable value, you can use the **echo** command and the variable you want to view. You also need to tell BASH that it's a variable by using the **$** in front of it. Here's an example:

```
$ echo $PATH

/usr/local/bin:/usr/bin:/bin:/usr/sbin:/sbin:/Applications/VMware
   Fusion.app/Contents/Public:/opt/X11/bin
```

To add a new value to the PATH variable, you can't just type **$PATH=/new_directory** because the operating system reads the environment variables only when the terminal session starts. To inform Linux that an environment variable needs to be updated, you use the **export** command. This command allows you to append your additional path to BASH and exists for the duration of the session. Make sure you add the **:** or **,** in front of your new value, depending on your operating system. The following example is for a Linux-style OS:

```
$ export PATH=$PATH:/Home/chrijack/bin
```

When you end your terminal session, the changes you made are not saved. To retain the changes, you need to write the path statement to your **.bashrc** (or **.zshrc** if using Z shell) profile settings. Anything written here will be available anytime you launch a terminal. You can simply copy, add the previous command to the end of the **.bashrc** with your favorite text editor, or use the following command:

```
$ echo "export PATH=$PATH:/Home/chrijack/bin" >> .bashrc
```

This addition becomes active only after you close your current session or force it to reload the variable. The **source** command can be used to reload the variables from the hidden configuration file **.bashrc**:

```
$ source ~/.bashrc
```

The following command does the same thing as the previous one because . is also an alias for the **source** command:

```
$ . ~/.bashrc
```

There are many more tricks you can uncover with BASH. You will get plenty of chances to use it as you study for the CCNAAUTO exam.

Software Version Control

The term *version control* is used to describe the process of saving various copies of a file or set of files in order to track changes made to those files. This description highlights how incredibly useful it is to the world of programming. Software version control (SVC) typically involves a database that stores current and historical versions of source code to allow multiple people or teams to work on it at the same time. If a mistake is made and you want to go back a revision (or to any previous version), SVC is the answer. You may also hear it called revision control or source control, but it all falls under the topic of software configuration management. Once you have used an SVC, you will quickly become a believer in its power and won't want to go back to using manual processes again.

If you have a team of developers working on a project, all writing new code or changing previously written code, losing those files could set you back weeks or months. Version control can protect your code by allowing changes to be checked in (through a process known as a *code commit*) to a hierarchical tree structure of folders with files in them. You often don't know what a developer might need to change or has changed, but the version control system does. Each check-in is tagged with who made the change and what the person changed within the code. Instead of using inefficient techniques such as file locking, a version control system handles concurrent check-ins, allowing two programmers to commit code at the same time.

Another aspect of a version control system is the ability to branch and merge code built independently. This is very useful if you are writing code on a part of an application that could conflict with another part written by another team. By creating a branch, you effectively create a separate work stream that has its own history and does not impact the main "trunk" of the code base. Once the code is written and any conflicts are resolved, the code from the branch can be merged back into the main trunk. Many application developers use this technique for new features or application revisions.

Can you write software without a version control system? Sure, but why would you? A lot of version control software options are available, many of them free, and it is good practice to always use a version control system to store your code. Git is one of the most commonly used version control systems today, and the 200-901 CCNAAUTO exam will test your knowledge of it, so the next section covers how to use Git.

Git

If you are working with version control software, chances are it is Git. A staggering number of companies use Git, which is free and open source. In 2005, Linus Torvalds (the father of Linux) created Git as an alternative to the SCM system BitKeeper, when the original owner of BitKeeper decided to stop allowing free use of the system for Linux kernel development. With no existing open-source options that would meet his needs, Torvalds created a distributed version control system and named it Git. Git was created to be fast and scalable, with a distributed workflow that could support the huge number of contributors to the Linux kernel. His creation was turned over to Junio Hamano in 2006, and it has become the most widely used source management system in the world.

> **Note**
>
> GitHub is not Git. GitHub is a cloud-based social networking platform for programmers that allows anyone to share and contribute to software projects (open source or private). While GitHub uses Git as its version control system underneath the graphical front end, it is not directly tied to the Git open-source project (much as a Linux distribution, such as Ubuntu or Fedora, uses the Linux kernel but is independently developed from it).

Understanding Git

Git is a distributed version control system built with scalability in mind. It uses a multi-tree structure, and if you look closely at the design, you see that it looks a lot like a file system. (Linus Torvalds is an operating system creator after all.) Git keeps track of three main structures, or trees (see Figure 2-8):

- **Local workspace:** This is where you store source code files, binaries, images, documentation, and whatever else you need.
- **Staging area:** This is an intermediary storage area for items to be synchronized (changes and new items).
- **Head, or local repository:** This is where you store all committed items.

Figure 2-8 *...Git Tree Structure*

Another very important concept with Git is the file lifecycle. Each file that you add to your working directory has a status attributed to it. This status determines how Git handles the file. Figure 2-9 shows the Git file status lifecycle, which includes the following statuses:

- **Untracked:** When you first create a file in a directory that Git is managing, it is given an untracked status. Git sees this file but does not perform any type of version control operations on it. For all intents and purposes, the file is invisible to the rest of the world. Some files, such as those containing settings or passwords or temporary files, may be stored in the working directory, but you may not want to include them in version control. If you want Git to start tracking a file, you have to explicitly tell it to do so with the **git add** command; once you do this, the status of the file changes to tracked.
- **Unmodified:** A tracked file in Git is included as part of the repository, and changes are watched. This status means Git is watching for any file changes that are made, but it doesn't see any yet.
- **Modified:** Whenever you add some code or make a change to the file, Git changes the status of the file to modified. Modified status is where Git sees that you are working on the file but you are not finished. You have to tell Git that you are ready to add a changed (modified) file to the index or staging area by issuing the **git add** command again.
- **Staged:** Once a changed file is added to the index, Git needs to be able to bundle up your changes and update the local repository. This process is called staging and is accomplished through **git commit**. At this point, your file status is moved back to the tracked status, and it stays there until you make changes to the file in the future and kick off the whole process once again.

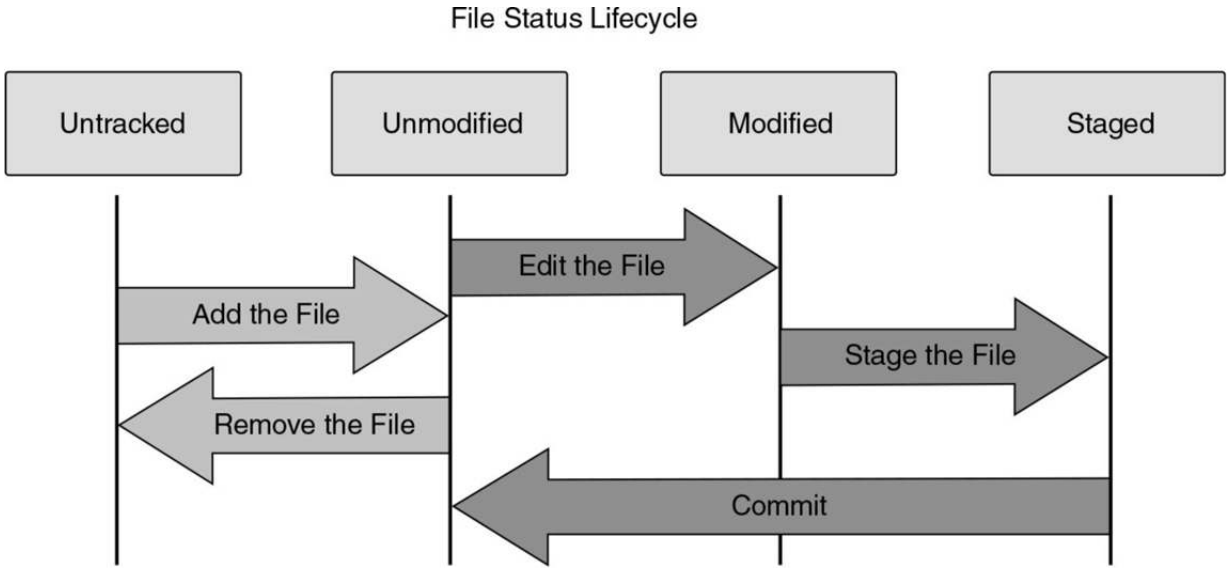


Figure 2-9 *...Git File Status Lifecycle*

If at any point you want to see the status of a file from your repository, you can use the extremely useful command **git status** to learn the status of each file in your local directory.

You can pull files and populate your working directory for a project that already exists by making a clone. Once you have done this, your working directory will be an exact match of what is stored in the repository. When you make changes to any source code or files, you can add your changes to the index, where they will sit in staging, waiting for you to finish all your changes or additions. The next step is to perform a commit and package up the changes for submission (or pushing) to the remote repository (usually a server somewhere local or on the Internet). This high-level process uses numerous commands

that are covered in the next section. If you understand Git's tree structure, figuring out what command you need is simple. Figure 2-10 shows the basic Git workflow.

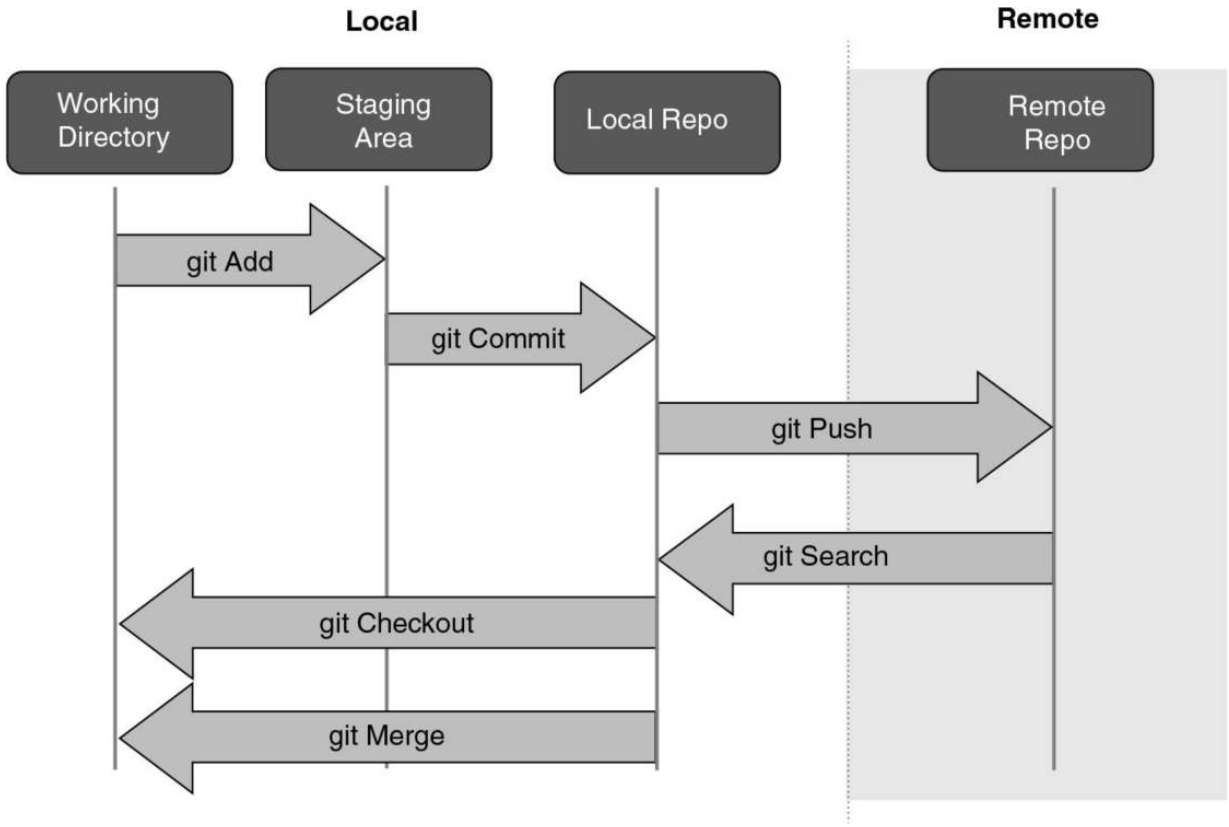


Figure 2-10 *...Git Workflow*

Using Git

Git may not come natively with your operating system. If you are running a Linux variation, you probably already have it. For Mac and Windows you need to install it. You can go to the main distribution website (https://git-scm.com) and download builds for your operating system directly. You can install the command-line version of Git and start using and practicing the commands discussed in this section. There are also GUI-based Git clients, but for the purposes of the CCNAAUTO exam, you should focus your efforts on the command line. Git commands come in two different flavors. The standard user-friendly commands are called "porcelain," and the more complicated inner workings of Git manipulating commands are called "plumbing." At its core, Git is a content-addressable file system. The version control system part was layered on top to make it easier to use. For the CCNAAUTO exam, you need to know your way around Git at a functional level (by using the porcelain). There is a significant amount of manipulation you can do with Git at the plumbing level. Most of the plumbing commands and tools are not ones you will be using on a regular basis and are not covered on the exam.

Cloning/Initiating Repositories

Git operates on a number of processes that enable it to do its magic. The first of these processes involves defining a local repository by using either **git clone** or **git init**. The **git clone** command has the following syntax:

```
git clone (url to repository) (directory to clone to)
```

If there is an existing repository you are planning to start working on, like one from GitHub that you like, you use **git clone**. This command duplicates an existing Git project from the URL provided into your current directory with the name of the repository as the directory name. You can also specify a different name with a command-line option. Example 2-3 shows an example of cloning a repository and listing the files in the newly created local repository.

Example 2-3 *...Cloning a GitHub Repository*

```
#git clone https://github.com/CiscoDevNet/pyats-coding-101.git
Cloning into 'pyats-coding-101'...
remote: Enumerating objects: 71, done.
remote: Total 71 (delta 0), reused 0 (delta 0), pack-reused 71
Unpacking objects: 100% (71/71), done.
#cd pyats-coding-101
#pyats-coding-101 git:(main) ls
COPYRIGHT                coding-102-parsers
LICENSE                     coding-103-yaml
README.md                coding-201-advanced-parsers
coding-101-python
 git init (directory name)
```

To create a completely new repository, you need to create a directory. Luckily, **git init** can be supplied with a directory name as an option to do this all in one command:

```
#git init newrepo

Initialized empty Git repository in /Users/chrijack/Documents/GitHub/newrepo/.git/

#newrepo git:(main)
```

What you just created is an empty repository, and you need to add some files to it. By using the **touch** command, you can create an empty file. The following example shows how to view the new file in the repository with the directory (**ls**) command:

```
#newrepo git:(main) touch newfile

#newrepo git:(main) ls

newfile
```

Once the file is added, Git sees that there is something new, but it doesn't do anything with it at this point. If you type **git status**, you can see that Git identified the new file, but you have to issue another command to add it to index for Git to perform version control on it. Here's an example:

```
# git status

On branch main


No commits yet


Untracked files:

  (use "git add <file>..." to include in what will be committed)


 newfile


nothing added to commit but untracked files present (use "git add" to track)
```

Git is helpful and tells you that it sees the new file, but you need to do something else to enable version control and let Git know to start tracking it.

Adding and Removing Files

When you are finished making changes to files, you can add them to the index. Git knows to then start tracking changes for the files you identified. You can use the following commands to add files to an index:

- **git add . or -A:** Adds everything in the entire local workspace.
- **git add (*filename*):** Adds a single file.

The **git add** command adds all new or deleted files and directories to the index. Why select an individual file instead of everything with the **.** or **-A** option? It comes down to being specific about what you are changing and adding to the index. If you accidently make a change to another file and commit everything, you might unintentionally make a change to your code and then have to do a rollback. Being specific is always safest. You can use the following commands to add the file newfile to the Git index (in a process known as *staging*):

```
# git add newfile

# git status

On branch main

No commits yet

Changes to be committed:

  (use "git rm --cached <file>..." to unstage)


  new file: newfile
```

Removing files and directories from Git is not as simple as just deleting them from the directory itself. If you just use file system commands to remove files, you may create headaches as the index can become confused. You can remove files and directories from Git, but it requires an extra step of adding the file deletion to the index (which sounds counterintuitive, right?). The best way is to use the **git rm** command, which has the following syntax:

```
git rm (-r) (-f) (folder/file.py)
```

This command removes a file or directory and syncs it with the index in one step. If you want to remove a directory that is not empty or has subdirectories, you can use the **-r** option to remove recursively. In addition, if you add a file to Git and then decide that you want to remove it, you need to use the **-f** option to force removal from the index. This is required only if you haven't committed the changes to the local repository. Here is an example:

```
# touch removeme.py

# git add .

# ls

newfile removeme.py

# git rm -f removeme.py

rm 'removeme.py'
```

git mv is the command you use to move or rename a file, directory, or symbolic link. It has the following syntax:

```
git mv (-f) (source) (destination)
```

For this command you supply a source argument and a destination argument to indicate which file or directory you want to change and where you want to move it. (Moving in this case is considered the same as renaming.) Keep in mind that when you use this command, it also updates the index at the same time, so there is no need to issue **git add** to add the change to Git. You can use the **-f** argument if you are trying to overwrite an existing file or directory where the same target exists. The following example shows how to change a filename in the same directory:

```
# ls

oldfile.py

# git mv oldfile.py newfile.py

# ls

newfile.py
```

Committing Files

When you commit a file, you move it from the index or staging area to the local copy of the repository. Git doesn't send entire updates; it sends just changes. The **commit** command is used to bundle up those changes to be synchronized with the local repository. The command is simple, but you can specify a lot of

options and tweaks. In its simplest form, you just need to type **git commit**. This command has the following syntax:

```
git commit [-a] [-m] <”your commit message”>
```

The **-a** option tells Git to add any changes you make to your files to the index. It’s a quick shortcut instead of using **git add -A**, but it works only for files that have been added at some point before in their history; new files need to be explicitly added to Git tracking. For every commit, you will need to enter some text about what changed. If you omit the **-m** option, Git automatically launches a text editor (such as vi, which is the default on Linux and Mac) to allow you to type in the text for your commit message. This is an opportunity to describe the changes you made so others know what you did. It’s tempting to type in something silly like “update” or “new change for a quick commit,” but don’t fall into that trap. Think about the rest of your team. Here is an example of the **commit** command in action:

```
# git commit -a -m “bug fix 21324 and 23421”

[main e1fec3d] bug fix 21324 and 23421

 1 file changed, 0 insertions(+), 0 deletions(-)

 delete mode 100644 newfile
```

> **Note**
>
> As a good practice, use the first 50 characters of the commit message as a title for the commit followed by a blank line and a more detailed explanation of the commit. This title can be used throughout Git to automate notifications such as sending an email update on a new commit with the title as the subject line and the detailed message as the body.

Conventional Commits

You’ll see a popular convention called Conventional Commits used in a lot of modern projects. The idea is to tag every commit with what kind of change it is. A feature, a bug fix, docs, refactor, and so on. The format is dead simple: a type, an optional scope in parens, a colon, and your description.

The types you’ll see most often are:

- feat: a new feature
- fix: a bug fix
- docs: documentation-only changes
- chore: maintenance that doesn’t change application behavior (build scripts, dependency updates, and so on)
- refactor: a code change that neither fixes a bug nor adds a feature
- test: adding or fixing tests
- BREAKING CHANGE: a footer that flags a backward-incompatible change

Here’s what one looks like:

```
feat(api): add pagination to /devices endpoint
```

Or with a longer body and a breaking change footer:

```
refactor(auth): unify token validation across services

Replaces the per-service token parsers with a single shared
implementation in the auth library.

BREAKING CHANGE: services must upgrade to auth-lib 3.0+
```

So why bother? A few tools (semantic-release, release-please, standard-version) actually read these messages and use them to generate changelogs and figure out semantic version numbers automatically. Even if you don't use those tools, your git log becomes much easier to scan when every commit's intent is obvious from the first word. A lot of open source projects now require Conventional Commits, enforced through pre-commit hooks or CI checks.

Pushing and Pulling Files

Up until this point in the chapter, you have seen how Git operates on your local computer. Many people use Git in just this way, as a local version control system to track documents and files. Its real power, however, is in its distributed architecture, which enables teams from around the globe to come together and collaborate on projects.

In order to allow Git to use a remote repository, you have to configure Git with some information so that it can find it. When you use the command **git clone** on a repository, Git automatically adds the remote repository connection information via the URL entered with the **clone** command.

When using the **git init** command, however, you need to make sure that you enter the information to find the remote location for the server with the **git remote add** command, which has the following syntax:

```
git remote add (name) (url)
```

git remote -v can be used to show which remote repository is configured. The following example shows how to add a remote repository and then display what is configured:

```
# git remote add origin https://github.com/chrijack/devnetccna.git

# git remote -v

origin https://github.com/chrijack/devnetccna.git (fetch)

origin https://github.com/chrijack/devnetccna.git (push)
```

What if you make a mistake or want to remove remote tracking of your repository? This can easily be done with the **git remote rm** command, which has the following syntax:

```
git remote rm (name)
```

Here is an example of this command in action:

```
# git remote rm origin

# git remote -v
```

In order for your code to be shared with the rest of your team or with the rest of the world, you have to tell Git to sync your local repository to the remote repository (on a shared server or service like GitHub). The command **git push**, which has the following syntax, is useful in this case:

```
git push (remotename) (branchname)
```

This command needs a remote name, which is an alias used to identify the remote repository. It is common to use the name origin, which is the default if a different name is not supplied. In addition, you can reference a branch name with **git push** in order to store your files in a separately tracked branch from the main repository. (You can think of this as a repository within a repository.) The sole purpose of the **git push** command is to transfer your files and any updates to your Git server. The following is an example of the **git push** command in use:

```
# git push origin main

Enumerating objects: 3, done.

Counting objects: 100% (3/3), done.

Writing objects: 100% (3/3), 210 bytes | 210.00 KiB/s, done.

Total 3 (delta 0), reused 0 (delta 0)

To https://github.com/chrijack/devnetccna.git

 * [new branch] main -> main

Branch 'main' set up to track remote branch 'main' from 'origin'.
```

The command **git pull** syncs any changes that are on the remote repository and brings your local repository up to the same level as the remote one. It has the following syntax:

```
git pull (remotename) (branchname)
```

Whenever you begin to work with Git, one of the first commands you want to issue is **pull** so you can get the latest code from the remote repository and work with the latest version of code from the main repository. **git pull** does two things: fetches the latest version of the remote main repository and merges it into the local repository. If there are conflicts, they are handled just as they would be if you issued the **git merge** command, which is covered shortly. Example 2-4 shows an example of using the **git pull** command.

Example 2-4 *...**git pull** Command*

```
# git pull origin main
remote: Enumerating objects: 9, done.
remote: Counting objects: 100% (9/9), done.
remote: Compressing objects: 100% (8/8), done.
remote: Total 8 (delta 1), reused 0 (delta 0), pack-reused 0
Unpacking objects: 100% (8/8), done.
From https://github.com/chrijack/devnetccna
 * branch            main     -> FETCH_HEAD
   8eb16e3..40aaf1a  main     -> origin/main
Updating 8eb16e3..40aaf1a
Fast-forward
2README.md                          |   3 +++
Picture1.png                        | Bin 0 -> 83650 bytes
Picture2.jpg                        | Bin 0 -> 25895 bytes
Picture3.png                        | Bin 0 -> 44064 bytes
 4 files changed, 3 insertions(+)
 create mode 100644 2README.md
 create mode 100644 Picture1.png
 create mode 100644 Picture2.jpg
 create mode 100644 Picture3.png
```

Working with Branches

Branches are an important workflow in software development. Say you want to add a new feature to your software or want to fix a bug. You can create a branch in order to add a separate development workspace for your project and prevent changes from destabilizing the main project (the main branch in Git). Remember that Git keeps a running history of every commit you make. This history (called a *snapshot* in Git terminology) details all the changes to the software over time and ensures the integrity of this record by applying an SHA-1 hash. This hash is a 40-character string that is tied to each and every commit. Example 2-5 shows an example of three commits with a hash, displayed using the **git log** command.

> **Note**
>
> Git used to default to master as the branch name. Since 2020, Git, GitHub, and GitLab all default to main instead. The two work exactly the same way; this book uses main throughout. You'll still see master in older repos.

There's a real downside to long-lived feature branches: the longer they live, the more they drift away from main, and the worse the eventual merge gets. An alternative approach called trunk-based development flips this around. Developers commit small, frequent changes straight to main, or use feature branches that live for hours or maybe a day or two, not weeks. Google, Facebook, and most other high-velocity engineering teams operate this way, and DORA's annual State of DevOps research consistently finds trunk-based development to be one of the practices that separates elite performers from the rest. It does take some discipline to make it work. You need feature flags so you can ship code without users seeing it yet, and solid CI to catch problems quickly. If you're shipping a versioned product on a release cadence (a library, a mobile app), Git Flow with longer-lived branches still makes more sense. If you're shipping continuously to production, trunk-based is usually the better fit.

Example 2-5 *...**git log** Command Output*

```
#git log

commit 40aaf1af65ae7226311a01209b62ddf7f4ef88c2 (HEAD -> main, origin/main)
Author: Chris Jackson <chrijack@cisco.com>
Date:  Sat Apr 25 00:00:34 2026 -0500

    Add files via upload

commit 1a9db03479a69209bf722b21d8ec50f94d727e7d
Author: Chris Jackson <chrijack@cisco.com>
Date:  Fri Apr 24 23:59:55 2026 -0500

    Rename README.md to 2README.md

commit 8eb16e3b9122182592815fa1cc029493967c3bca
Author: Chris Jackson <chrijack@me.com>
Date:  Fri Apr 24 20:03:32 2026 -0500

    first commit
```

Notice that the first entry is the current commit state, as it is referenced by HEAD. The other entries show the chronological history of the commits. Figure 2-11 shows a visual representation of this simple three-step commit history; for brevity, only the first four values of the hash are used.

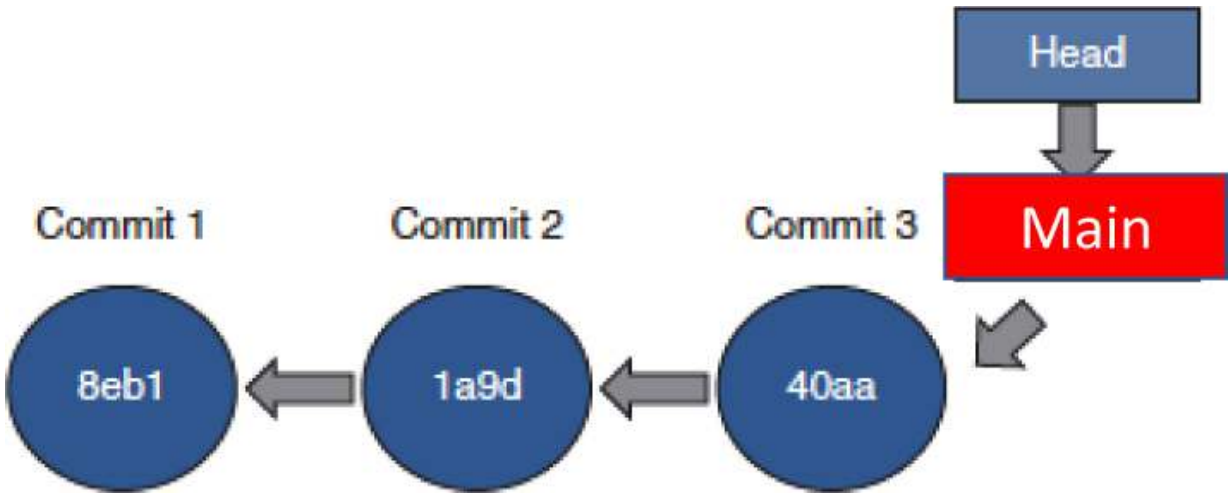


Figure 2-11 *...Git Commit History*

To add a Git branch, you simply issue the **git branch** command and supply the new branch with a name, using the following syntax:

```
git branch (-d) <branchname> [commit]
```

You can alternatively specify a commit identified by a tag or commit hash if you want to access a previous commit from the branch history. By default, Git selects the latest commit. In addition, you can delete a branch when you no longer need it but using the **-d** argument. The following example shows how to create a branch and display the current branches with the **git branch** command with no argument:

```
# git branch newfeature

# git branch


* main

 newfeature
```

The * next to main shows that the branch you are currently in is still main, but you now have a new branch named **newfeature**. Git simply creates a pointer to the latest commit and uses that commit as the start for the new branch. Figure 2-12 shows a visual representation of this change.

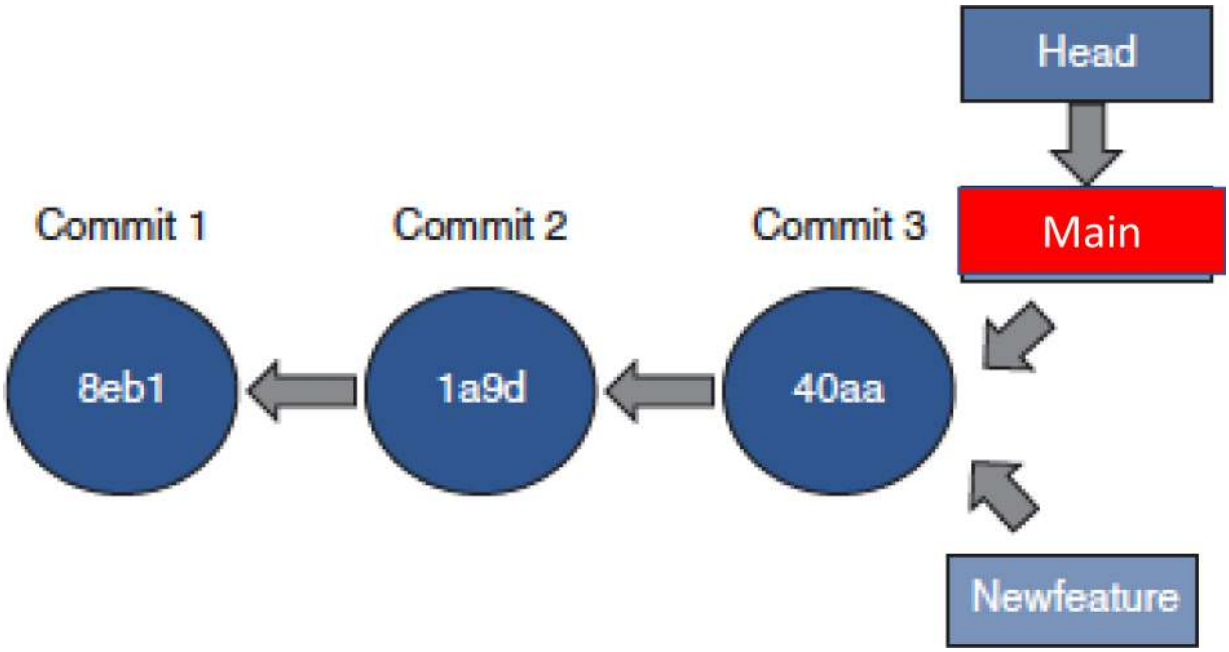


Figure 2-12 *...Adding a Branch*

In order to move to the new branch and change your working directory, you have to use the **git checkout** command, which has the following syntax:

```
git checkout [-b] (branchname or commit)
```

The **-b** argument is useful for combining the **git branch** command with the checkout function and saves a bit of typing by creating the branch and checking it out (switching to it) all at the same time. This example moves Head on your local machine to the new branch, as shown in Figure 2-13:

```
#git checkout newfeature

Switched to branch 'newfeature'
```

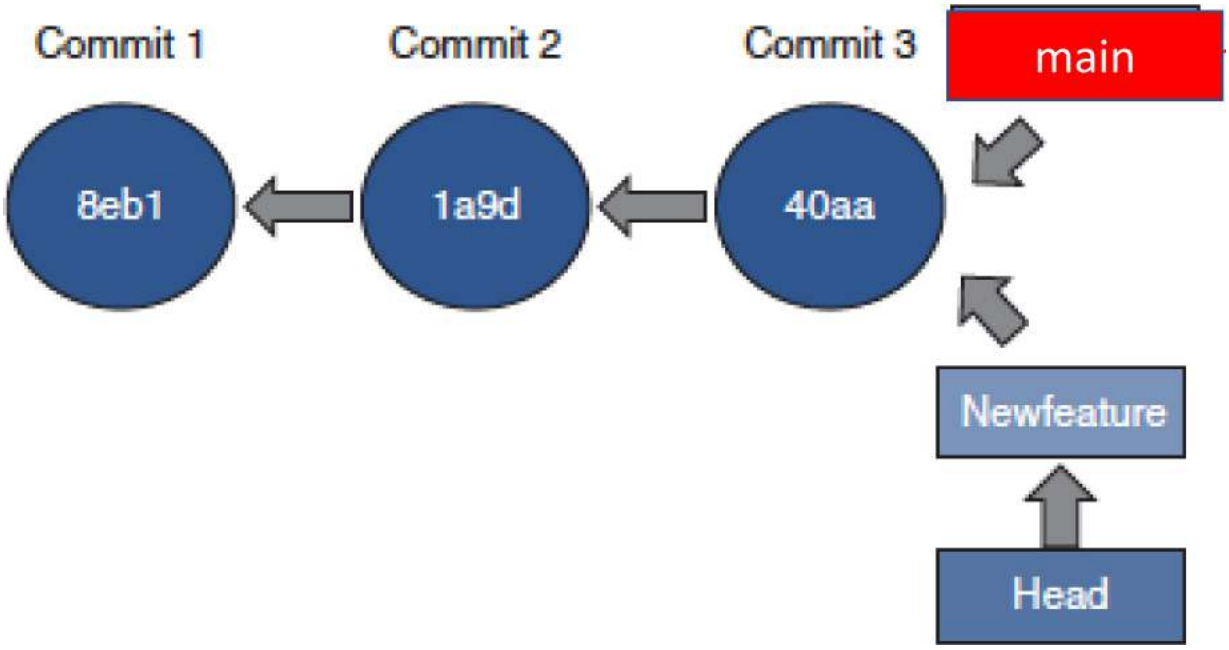


Figure 2-13 *...Changing Head to a New Branch*

Now you have a separate workspace where you can build your feature. At this point, you will want to perform a **git push** to sync your changes to the remote repository. When the work is finished on the branch, you can merge it back into the main code base and then delete the branch by using the command **git branch -d (branchname)**.

Merging Branches

The merge process in Git is used to handle the combining of multiple branches into one. The **git merge** command is used to make this easier on the user and provide a simple way to manage the changes. It has the following syntax:

```
git merge (branch to merge with current)
```

To understand the merge process, it helps to take a look at your two branches. Figure 2-14 shows all of the commits that have taken place as part of the feature build. In addition, you can see other commits that have also occurred on the main branch.

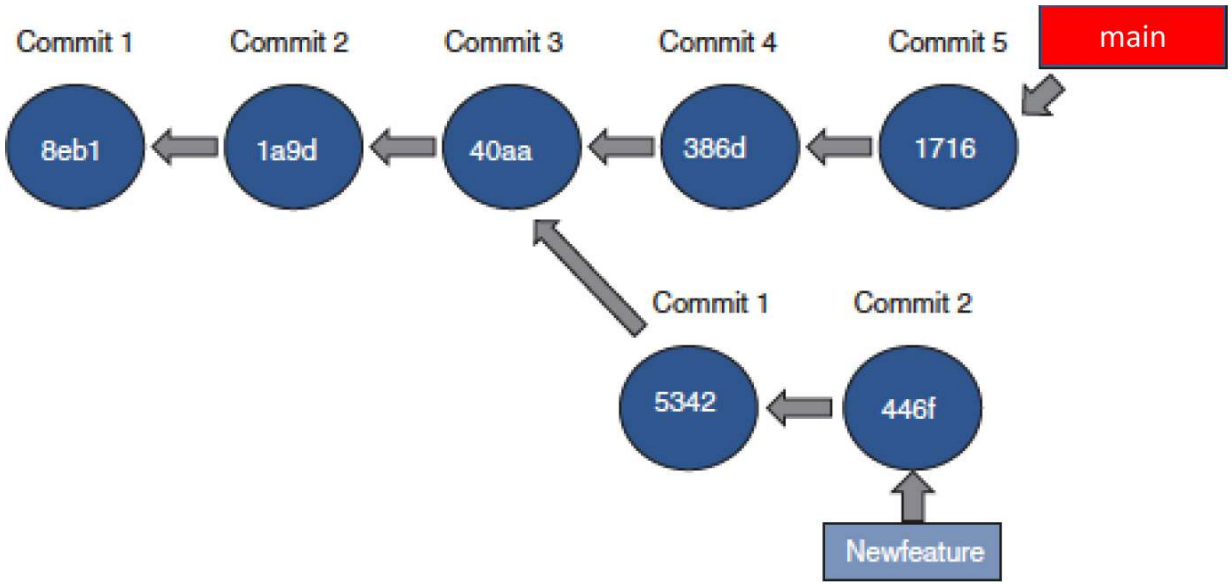


Figure 2-14 *...Two Branches with Commits*

In order to get the two branches merged, Git has to compare all the changes that have occurred in the two branches. You have a text file that exists in both the main branch and the newfeature branch, and for simplicity's sake, there are just a couple of lines of text. Figure 2-15 shows the main branch text file.

Figure 2-15 *...Main Branch Text File*

In the newfeature branch, this text file has been modified with some new feature code. Figure 2-16 shows a simple change made to the text file.

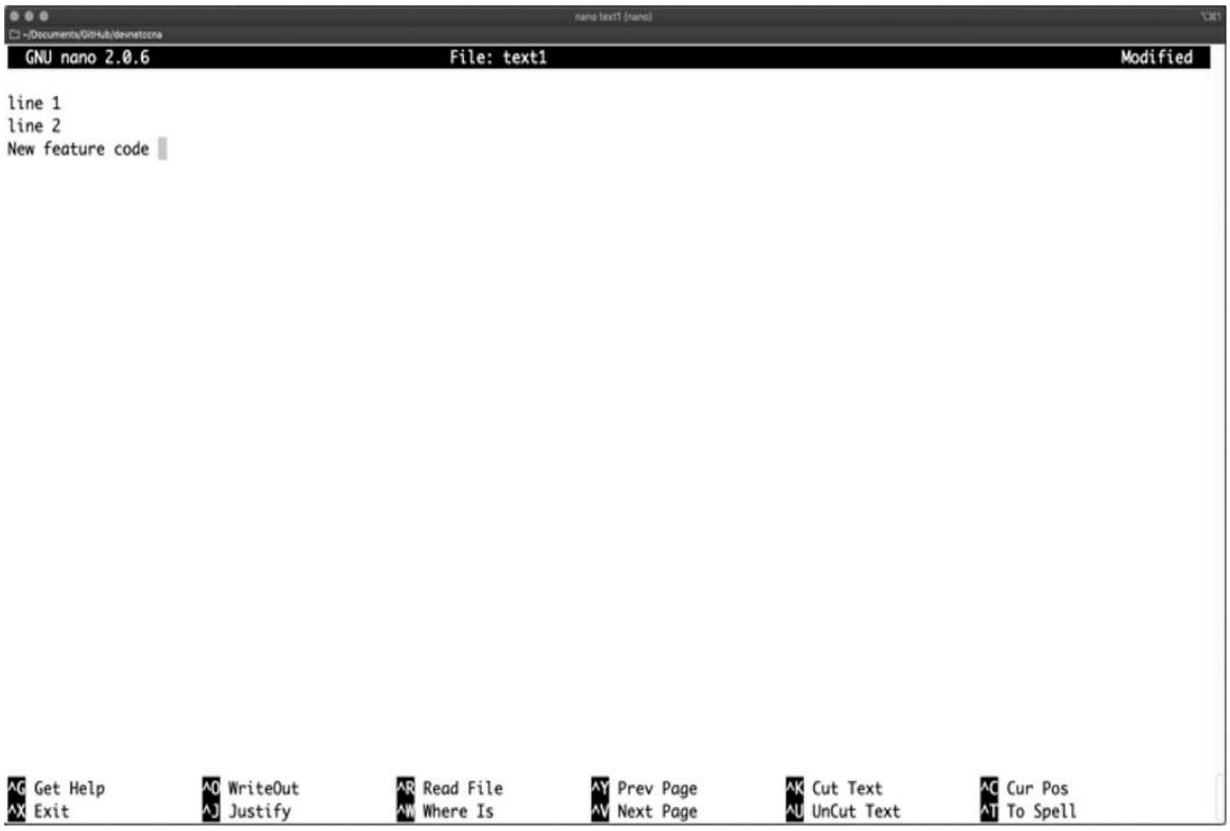


Figure 2-16 *...Changes to the Text File in the newfeature Branch*

On the newfeature branch, you can issue the following commands to add the changes to the index and then commit the change:

```
#git add .

#git commit -a -m "new feature"
```

Now the branch is synced with the new changes, and you can switch back to the main branch with the following command:

```
#git checkout main

Switched to branch 'main'
```

From the main branch, you can then issue the **git merge** command and identify the branch to merge with (in this case, the newfeature branch):

```
# git merge newfeature

Updating 77f786a..dd6bce5

Fast-forward

 text1 | 1 +

 1 file changed, 1 insertion(+)
```

In this very simple example, no changes were made to the main branch, and Git was able to automatically merge the two branches and create a new combined commit that had the new content from the newfeature branch. Notice that the output above says "Fast-forward"; this refers to updating past the changes in the branch, which is much like fast-forwarding through the boring parts of a movie. At this point, you can delete the branch newfeature, as the code in it has been moved to main. Figure 2-17 illustrates how this is done: Git creates a new commit that has two sources.

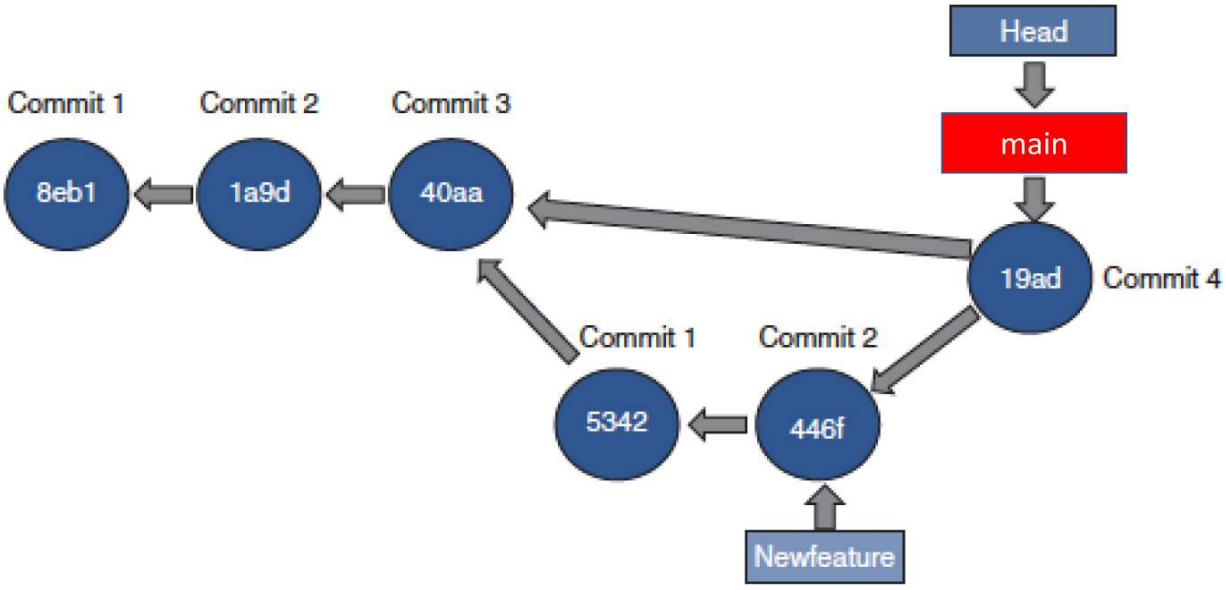


Figure 2-17 *...Git Merge Between Two Branches*

Handling Conflicts

Merging branches is a very useful capability, but what happens if the same file is edited by two different developers? You can have a conflict in terms of which change takes precedence. Git attempts to handle merging automatically, but where there is conflict, Git relies on human intervention to decide what to keep. In the previous example, if there had been changes made to text1 in both the main branch and the newfeature branch, you would have seen the following message after using the command **git merge**:

```
#git merge newfeature

Auto-merging text1

CONFLICT (content): Merge conflict in text1

Automatic merge failed; fix conflicts and then commit the result.
```

In addition, text1 would look as shown in Figure 2-18 (which shows the conflicting merge).

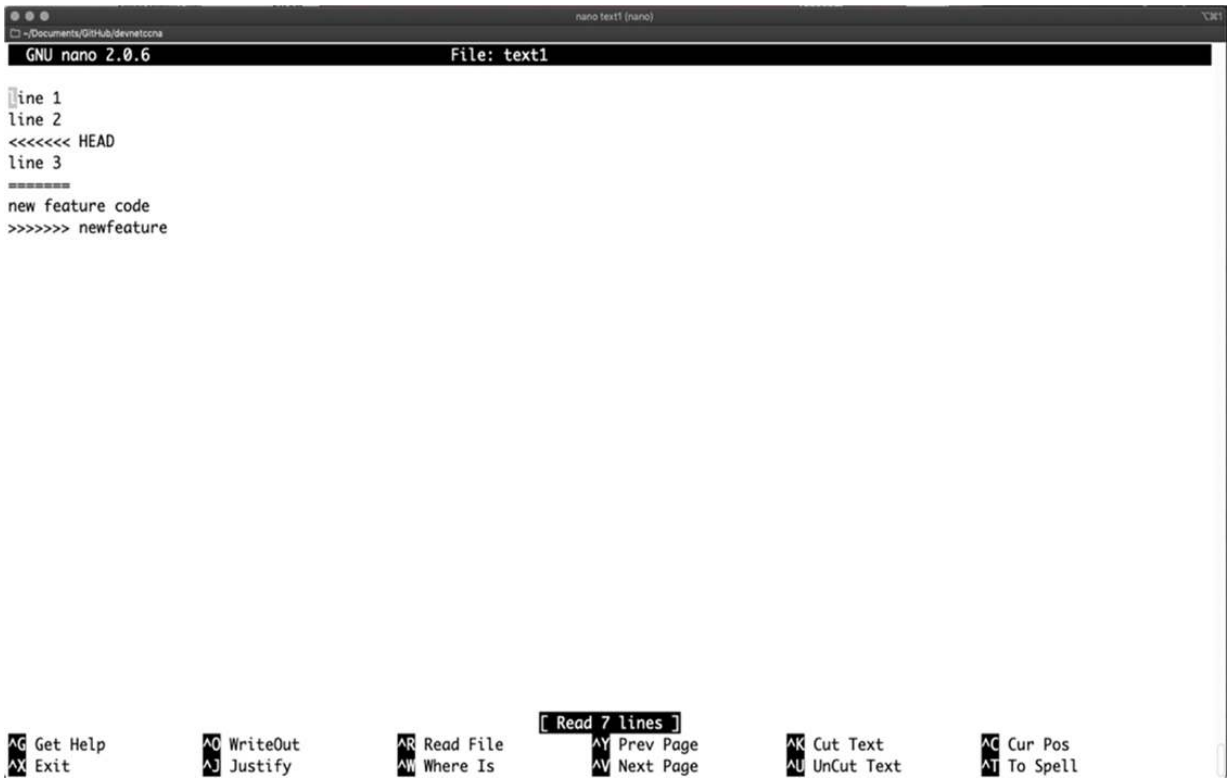


Figure 2-18 *...Git Conflicting Merge*

Git shows you that "line 3" was added to text1 on the main branch and "new feature code" was added to text1 on the newfeature branch. Git is letting you delete one or keep both. You can simply edit the file, remove the parts that Git added to highlight the differences, and save the file. Then you can use **git add** to index your changes and **git commit** to save to the local repository, as in the following example:

```
#git add .

#git commit -m "merge conflict fixed"

[main fe8f42d] merge conflict fixed
```

Comparing Commits with diff

The **diff** command is one of the most powerful Git tools. It allows you to compare files and text to see which you want to use if there are multiple options. The ability to compare specific commits in Git makes it easier to know what to keep and what to discard between two similar versions of code.

The **diff** command takes two sets of inputs and outputs the differences or changes between them. This is its syntax:

```
git diff [--stat] [branchname or commit]
```

git diff looks at the history of your commits, individual files, branches, and other Git resources. It's a very useful tool for troubleshooting issues as well as comparing code between commits. It has a lot of options and command-line parameters, which makes it a bit of a Swiss Army knife in terms of functionality. One of the most useful functions of **diff** is to be able to see the differences between the three Git tree structures. The following are variations of the **git diff** command that you can use:

- **git diff:** This command highlights the differences between your working directory and the index (that is, what isn't yet staged).
- **git diff --cached:** This command shows any changes between the index and your last commit.
- **git diff HEAD:** This command shows the differences between your most recent commit and your current working directory. It is very useful for seeing what will happen with your next commit.
- git diff <commit1> <commit2>: This command shows the differences between two specific commits. Use it to compare any two points in your project history by their commit hashes.

The following is an example of executing **git diff --cached** after **text2** is added to the index:

```
#git diff --cached

diff --git a/text2 b/text2

new file mode 100644

index 0000000..b9997e5

--- /dev/null

+++ b/text2

@@ -0,0 +1 @@

+new bit of code
```

git diff identified the new file addition and shows the a/b comparison. Since this is a new file, there is nothing to compare it with, so you see **--- /dev/null** as the *a* comparison. In the *b* comparison, you see **+++ b/text2**, which shows the addition of the new file, followed by stacks on what was different. Since there was no file before, you see **-0,0** and **+1**. (The **+** and **-** simply denote which of the two versions you are comparing. It is not actually a **-0**, which would be impossible.) The last line shows the text that was added to the new file. This is a very simple example with one line of code. In a big file, you might see a significant amount of text.

Another very useful capability of **git diff** is to compare branches. By using **git diff** (branchname), you can see the differences between a file in the branch you are currently in and one that you supply as an argument. The following compares text1 between the branches main and newfeature, where you can see that line 3 is present on newfeature branch's text1 file:

```
#git diff newfeature text1

 diff --git a/text1 b/text1

index 45c2489..ba0a07d 100644

--- a/text1

+++ b/text1

@@ -1,3 +1,4 @@

 line 1

 line 2

+line 3

 new feature code
```

This section has covered quite a bit of the syntax and commands that you will see on a regular basis when working with Git. You need to spend some time working with these commands on your local machine and become familiar with them so you will be ready for the 200-901 CCNAAUTO exam. Make sure you are using resources such as Git documentation on any command you don't understand or for which you want to get deeper insight. All of these commands have a tremendous amount of depth for you to explore.

Conducting Code Review

Every good author needs an editor. This book wouldn't have been even half as understandable if it hadn't been for the fact that we had other people check our work for comprehension and technical accuracy. Why should code you write be treated any differently? The intent behind a code review process is to take good code and make it better by showing it to others and having them critique it and look for potential errors. When you develop software, the vast majority of your time is spent by yourself—just you and the keyboard. Sometimes when you are this close to a software project, you miss errors or use ineffective coding techniques; a simple code review can quickly uncover such issues.

Beyond the aspects mentioned above, why should you conduct code reviews? The following are a few common benefits of code review:

- It helps you create higher-quality software.
- It enables your team to be more cohesive and deliver software projects on time.
- It can help you find more defects and inefficient code that unit tests and functional tests might miss, making your software more reliable.

There are many ways to conduct code reviews. Some organizations use specialized applications such as Gerrit, while most teams use the pull request (or merge request) workflow built into GitHub, GitLab, and Bitbucket, where reviewers comment inline on proposed changes before code is merged. Others conduct reviews as if they were professors grading college papers. Whatever process you use, the following are some good practices to help make your code review effective:

- Use a code review checklist that includes organization-specific practices (naming conventions, security, class structures, and so on) and any areas that need special consideration. The goal is to have a repeatable process that is followed by everyone.
- Review the code, not the person who wrote it. Avoid being robotic and harsh so you don't hurt people's feeling and discourage them. The goal is better code, not disgruntled employees.
- Keep in mind that code review is a gift. No one is calling your baby ugly. Check your ego at the door and listen; the feedback you receive will make you a better coder in the long run.
- Make sure the changes recommended are committed back into the code base. You should also share findings back to the organization so that everyone can learn from mistakes and improve their techniques.

Most teams catch the boring quality issues before code ever reaches a pull request by using pre-commit hooks. These are short scripts that run every time you try to commit. They can lint your code (Ruff, ESLint), format it (Black, Prettier), scan for accidentally committed secrets (gitleaks, detect-secrets), or check that your commit message follows whatever convention you're using. The pre-commit framework at pre-commit.com makes wiring all this up a single config-file change. The payoff is that reviewers stop having to nag people about formatting or missing semicolons or leaked secrets. They can focus on whether the code is doing the right thing in the right way.

If you want a single yardstick for how well your team is shipping software, the four DORA metrics are the most widely used. They came out of Google's DevOps Research and Assessment team and have become the standard measurement framework in the industry. The four are: how often you deploy (deployment frequency), how long it takes a commit to reach production (lead time for changes), what percentage of your deployments break something (change failure rate), and how fast you can recover when something does break (time to restore service). The benchmarks are pretty stark. Elite teams deploy multiple times a day with lead times under an hour. Low performers measure in weeks or months. The way you do code review (fast turnaround, small focused PRs, real automation) affects all four of these directly.

All the major Git hosts (GitHub, GitLab, Bitbucket) let you protect important branches with rules. The common ones: no direct pushes (everything goes through a PR), a minimum number of approving reviews before you can merge, all CI checks must pass, and no force-pushes allowed. On top of that, a CODEOWNERS file in your repo can map file paths to required reviewers automatically. Touch the auth code, and the security team gets pulled into the PR. Touch infra code, and the platform team is required. With all this enabled, code review stops being a polite suggestion and becomes part of how merging actually works. Skipping it isn't an option anymore.

The Pull Request Workflow

Knowing the Git commands is one thing. Knowing how teams actually use them is another. Most teams don't push changes straight to main. Almost everyone uses some flavor of the pull request workflow (or merge request, on GitLab). It builds the entire collaboration story around the commands you've already seen.

Here's the basic flow:

Step 1. Create a feature branch off main: git checkout -b feature/short-description.

Step 2. Commit your changes locally, then push the branch up: git push origin feature/short-description.

Step 3. Open a pull request on GitHub, GitLab, or Bitbucket. Describe what changed and why, and link any related issues.

Step 4. Reviewers comment on specific lines, request changes, or approve. Automated checks (CI builds, tests, linters) run on every push you make to the branch.

Address feedback by pushing more commits to the same branch. The PR updates automatically.

Step 6. Once you have approvals and all checks pass, merge the PR into main. The feature branch usually gets deleted right after.

The big advantage of this pattern is that code review stops being a calendar event ("we'll review on Friday") and becomes part of how code lands in the first place. Nothing reaches main without somebody else looking at it. As a bonus, every PR is a permanent record of what changed, who looked at it, and what tests ran. That's useful when you're trying to figure out why something was done six months later.

Exam Preparation Tasks

As mentioned in the section "How to Use This Book" in the Introduction, you have a couple of choices for exam preparation: the exercises here, Chapter 19, "Final Preparation," and the exam simulation questions on the companion website.

Review All Key Topics

Review the most important topics in this chapter, noted with the Key Topic icon in the outer margin of the page. Table 2-2 lists these key topics and the page number on which each is found.

Table 2-2 *...Key Topics*

Key Topic Element	Description	Page Number
Paragraph	Waterfall	
Paragraph	Lean	
Paragraph	Agile	
Paragraph	Model-View-Controller (MVC) pattern	
Section	Observer Pattern	
Paragraph	Getting to Know BASH	
Paragraph	Software version control	
Section	Using Git	
Section	Conducting Code Review	

Define Key Terms

Define the following key terms from this chapter and check your answers in the glossary:

Software Development Lifecycle (SDLC)
Waterfall
Lean
Agile
Model-View-Controller (MVC)
Observer
software version control
Git
GitHub
repository
staging
index
local workspace

Chapter 3. Introduction to Python

This chapter covers the following topics:

- **Getting Started with Python:** This section covers what you need to know when using Python on your local machine.
- **Understanding Python Syntax:** This section describes the basic Python syntax and command structure.
- **Data Types and Variables:** This section describes the various types of data you need to interact with when coding.
- **Input and Output:** This section describes how to get input from a user and print out results to the terminal.
- **Flow Control with Conditionals and Loops:** This section discusses adding logic to your code with conditionals and loops.

Python is an easy language that anyone can learn quickly. It has become the de facto language for custom infrastructure automation. Thanks to its English-like command structure, readability, and simple programming syntax, you will find that you can accomplish your goals more quickly with Python than with languages such as C or Java. While you are not expected to be an expert in Python for the 200-901 CCNAAUTO exam, you do need to be fluent enough to understand what is going on in a sample of Python code. You also need to be able to construct Python code by using samples from DevNet and GitHub to interact with Cisco products. While the next few chapters are not intended to replace a deep dive into Python programming, they serve as a starting point for success on the exam.

"Do I Know This Already?" Quiz

The "Do I Know This Already?" quiz allows you to assess whether you should read this entire chapter thoroughly or jump to the "Exam Preparation Tasks" section. If you are in doubt about your answers to these questions or your own assessment of your knowledge of the topics, read the entire chapter. Table 3-1 lists the major headings in this chapter and their corresponding "Do I Know This Already?" quiz questions. You can find the answers in Appendix A, "Answers to the 'Do I Know This Already?' Quiz Questions."

Table 3-1 *"Do I Know This Already?" Section-to-Question Mapping*

Foundation Topics Section	Questions
Getting Started with Python	1, 2
Understanding Python Syntax	3, 4
Data Types and Variables	5, 6
Input and Output	7, 8
Flow Control with Conditionals and Loops	9, 10

Caution

The goal of self-assessment is to gauge your mastery of the topics in this chapter. If you do not know the answer to a question or are only partially sure of the answer, you should mark that question as wrong for purposes of self-assessment. Giving yourself credit for an answer that you correctly guess skews your self-assessment results and might provide you with a false sense of security.

1. What is the appropriate way to create a virtual environment for Python 3?

   a. python3 -virtual myvenv

   b. python3 virtual myvenv

   c. python3 -m vrt myvenv

   d. python3 -m venv myvenv

2. What command is used to install Python modules from PyPI?

   a. pip load packagename

   b. pip install packagename

   c. python3 -m pip install packagename

   d. python3 -t pip install packagename

3. What is the standard for indention in Python?

   a. One space for each block of code

   b. Four spaces for each block of code

   c. One tab for each block of code

   d. One tab and one space per block of code

4. How are comments in Python denoted?

   a. // on each line you want to make a comment

   b. # or ‘” quotation marks encompassing multiline comments

   c. /* comment */

   d. @$ comment %@

5. Which of the following are mutable data types? (Choose two.)

a. Lists

b. Dictionary

c. Integers

d. Tuples

6. Which of the following would create a dictionary?

a. a= (“ name”,”chris”,”age”,45)

b. a= dict()

c. a= [name, chris, age, 45]

d. a= {“ name”:”chris”, “age”: 45}

7. What data type does the **input()** function create when assigned to a variable?

a. List

b. Raw

c. String

d. An auto typed one

8. Which print statement is valid for Python 3?

a. print ‘hello world’

b. print(‘hello world”)

c. print(hello, world)

d. print(“‘hello world’”)

9. How do **if** statements operate?

a. If evaluates a variable against a condition to determine whether the condition is true.

b. If uses Boolean operators.

c. An if statement needs to end with :.

d. All of the above are correct.

10. Which statements are true about the **range()** function? (Choose two.)

a. The **range()** function iterates by one, starting at 0, up to but not including the number specified.

b. The **range()** function iterates by one, starting at 1, up to the number specified.

c. A **range()** function cannot count down, only up.

d. A **range()** function can count up or down, based on a positive or negative step value.

Foundation Topics

Getting Started with Python

Those from many engineering backgrounds are looking to integrate programming into their infrastructure. Maybe you are a hardcore computer science major and have been programming in multiple languages for years. You might be an infrastructure engineer who is strong in the ways of Cisco IOS and looking for new ways to operate in a diverse environment. You might even be a server expert who is familiar with Ansible or Terraform automation and are being asked to bring that automation knowledge to the

networking team. Regardless of your background or experience level, the CCNA Automation certification is designed to help you build the competency needed to be successful and give you a chance to prove what you have learned. If you haven't coded in years, or if the language that you currently program in isn't one that is very popular for infrastructure automation, where do you start? Python.

The Python language has become the most popular language in infrastructure automation because it is super easy to pick up and doesn't have all of the crazy syntax and structure that you see in languages like Java or C. It's based on the English language and is not only readable but extendable and powerful enough to be the one language you can count on to be able to get things accomplished in your day-to-day life. Those repetitive tasks that suck your productivity dry can be automated with just a few lines of code. Plus, due to the popularity of Python, millions of sample scripts provided by users like you as well as engineers at Cisco are free on GitHub for you to use and modify.

The software company TIOBE has published a list of the most popular programming languages each year for the past 10 years, and Python has consistently made the list. As of 2026, Python ranks number 1, and it has held the top position consistently for the past several years. In addition, a significant number of job postings reference Python programming skills as a requirement for successful candidates. Python is used in AI, machine learning, big data, robotics, security, penetration testing, and many other disciplines that are being transformed by automation. Needless to say, learning Python has become a differentiator for skilled engineers, and it is part of the core tool set for DevOps and cloud operational models.

The 200-901 CCNAAUTO exam is not a Python test per se. You will not be asked to answer esoteric questions about techniques and syntax that only a Python wizard would know. The exam ensures that you are competent enough with Python to know how to interact with Cisco hardware through APIs and to use Cisco software development kits and frameworks such as pyATS and Genie. It is a very good idea to continue learning Python and spend some time in either online courses or self-study via books focused on the Python language itself; you should also spend lots of time working through examples on DevNet at developer.cisco.com. This chapter and the several that follow provide a crash course in functional Python to get you going with the basics you need for success on the exam.

Many Linux distributions ship with Python 3 already installed, but Windows and macOS no longer include a current, supported Python by default. On Windows, the easiest path is the Microsoft Store (search for "Python") or the official installer from python.org. On macOS, head over to python.org for the graphical installer, or use Homebrew (brew install python) if you already use it. Apple does ship an older Python 3 with the Xcode Command Line Tools, but on macOS 26 Tahoe it's still Python 3.9, which reached end-of-life in October 2025. Install your own. The current stable is Python 3.12 or later, and the installation is quick. There are plenty of tutorials online if you get stuck.

> **Note**
>
> If you come across older Python 2.x code in legacy projects, expect noticeable differences from Python 3. The biggest things: print is a function in 3, the standard library was reorganized, and string handling changed. Apple stopped shipping Python 2.7 with macOS 12.3 in 2022, so you won't even find it on a current Mac. You'll learn more about common Python modules in Chapter 4, "Python Functions, Classes, and Modules."

The use of Python 3 has changed dramatically as support for the 2.x version ended in January 2020. The 3.x version came out in 2008 and is the one that you should be using today. Of course, this version issue is still a problem even within the 3.x train of Python and the corresponding modules you may want to use. To address this compatibility conundrum across different versions and modules, Python virtual environments have been created. Such an environment allows you to install a specific version of Python and packages to a separate directory structure. This way, you can ensure that the right modules are loaded, and your applications don't break when you upgrade your base Python installation. As of Python 3.3, there is native support for these virtual environments built into the Python distribution. You can code in Python without using virtual environments, but the minute you update modules or Python itself, you run

the risk of breaking your apps. A virtual environment allows you to lock in the components and modules you use for your app into a single "package," which is a good practice for building Python apps.

To use virtual environments, you launch Python 3 with the **-m** argument to run the **venv** module. You need to supply a name for your virtual environment, which will also become the directory name that will include all the parts of your virtual environment. Next, you need to activate the virtual environment by using the **source** command in Linux or Mac, as shown in this example. On Windows, you will need to run the activate batch file.

```
# MacOS or Linux

python3 -m venv myvenv

source myvenv/bin/activate



# Windows

C:\py -3 -m venv myvenv

C:\myvenv\Scripts\activate.bat
```

At this point, you will see your virtual environment name in parentheses at the beginning of your command prompt:

```
(myvenv)$
```

This indicates that you are running Python in your virtual environment. If you close the terminal, you have to reactivate the virtual environment if you want to run the code or add modules via **pip** (the Python module package manager).

To turn off the virtual environment, just type **deactivate** at the command prompt, and your normal command prompt returns, indicating that you are using your local system Python setup and not the virtual environment.

To install new modules for Python, you use **pip**, which pulls modules down from the PyPI repository. The command to load new modules is as follows:

```
pip install packagename
```

where *packagename* is the name of the package or module you want to install. You can go to pypi.org and search for interesting modules and check to see what others are using. There are more than 200,000 projects in the PyPI repository, so you are sure to find quite a few useful modules to experiment with. These modules extend Python functionality and contribute to its flexibility and heavy use today.

The **pip** command also offers a search function that allows you to query the package index:

```
pip search "search value"
```

The output includes the name, version, and a brief description of what the package does.

To install a specific version of a package, you can specify a version number or a minimum version so that you can get recent bug fixes:

```
pip install package==1.1.1. To install a specific version

pip install package>=1.0 To install a version greater than or equal to 1.0
```

When you download sample code, if there are package dependencies, there is usually a readme file that lists these requirements.

Using a requirements.txt file included with your code is another essential good practice. Such a file makes it simpler to get your Python environment ready to go as quickly as possible. If you have a requirements.txt file included with your code, it will give **pip** a set of packages that need to be installed, and you can issue this one command to get them loaded:

```
pip install -r requirements.txt
```

The requirements.txt file is just a list that maps Python package names to versions. Example 3-1 shows what it looks like.

Example 3-1 *Contents of requirements.txt*

```
ansible-core>=2.16
black>=24.0
flake8>=7.0
genie>=24.0
ipython>=8.0
napalm>=5.0
ncclient>=0.6.15
netmiko>=4.3
pyang>=2.6
pyats>=24.0
PyYAML>=6.0
requests>=2.31
urllib3>=2.0
virlutils>=0.9
xmltodict>=0.13
```

Modern Python projects increasingly use pyproject.toml (PEP 621) instead of (or alongside) requirements.txt. pyproject.toml is a single file that holds project metadata, dependencies, build configuration, and tool settings (Black, Ruff, mypy, pytest). requirements.txt still works and is fine for the examples in this book; pyproject.toml is what you'll see in most new projects.

As of 2025, uv (from Astral) has emerged as a fast modern alternative to pip and venv. Written in Rust, it installs packages 10–100× faster than pip and can manage virtual environments and Python versions in one tool. The standard pip and venv commands shown in this chapter still work everywhere; uv is worth trying once you're comfortable with the basics.

If you are building your own code and want to save the current modules configured in your virtual environment, you can use the **freeze** command and have it automatically populate the requirements.txt file:

```
pip freeze > requirements.txt
```

Understanding Python Syntax

The word *syntax* is often used to describe structure in a language, and in the case of programming syntax, is used in much the same way. Some programming languages are very strict about how you code, which can make it challenging to get something written. While Python is a looser language than some, it does have rules that should be followed to keep your code not only readable but functional. Keep in mind that Python was built as a language to enhance code readability and named after Monty Python (the British comedy troop) because the original architects of Python wanted to keep it fun and uncluttered. Python is best understood through its core philosophy (The Zen of Python):

- Beautiful is better than ugly.
- Explicit is better than implicit.
- Simple is better than complex.
- Complex is better than complicated.
- Readability counts.

Python is a scripted language, which means that any text editor can become a Python programming environment. Your choice of editor is completely up to your own preferences. The easiest way to get started with some Python code is to just enter **python3** at the command prompt (assuming that Python 3 is installed, of course) and use the interactive interpreter:

```
$ python3

>>> print("Savannah rules!")

Savannah rules!
```

For doing simple repetitive tasks, the interpreter is a quick interface to Python. For any program that you want to do a lot of editing, an editor is your best bet. Visual Studio Code (VS Code) is the most widely used editor for Python today, with Cursor and JetBrains PyCharm as strong alternatives. Any editor will do, but a strong ecosystem of extensions (like the official Python extension for VS Code) makes a particular editor much more useful. Figure 3-1 shows VS Code with a Python file open.

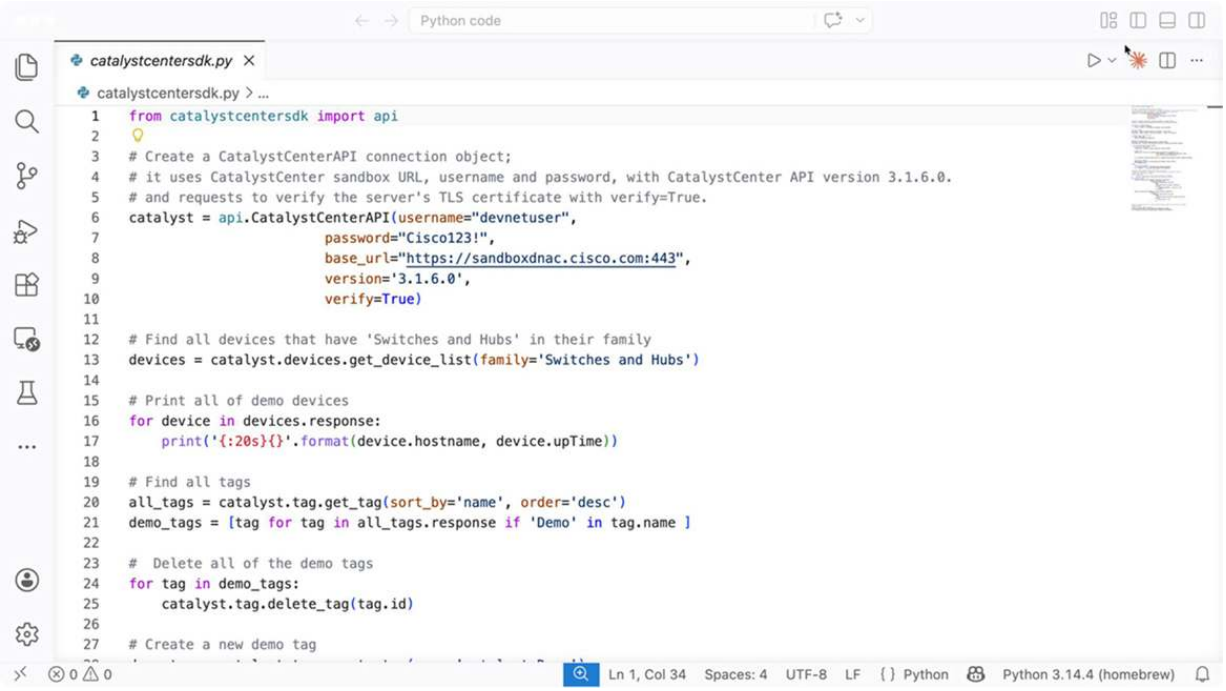


Figure 3-1 *Visual Studio Code Editor*

One aspect in which Python is different from other languages is that within Python code, whitespace matters. This can seem really weird and frustrating if you are coming from another language, such as Java or C, that uses curly braces or start/stop keywords; instead, Python uses indentation to separate blocks of code. This whitespace is not used just to make your code readable; rather, Python will not work without it. Here is an example of a simple loop that highlights why whitespace is so important:

```
>>> for kids in [“Caleb”, “Sydney”, “Savannah”]:

... print(“Clean your room,”, kids, “!”)

 File “<stdin>”, line 2

   print(“Clean your room,”, kids, “!”)

    ^

IndentationError: expected an indented block
```

This code will generate a syntax error the minute you try to run it. Python is expecting to see indentation on the line after the **:** . If you insert four spaces before the **print()** statement, the code works:

```
>>> for kids in [“Caleb”, “Sydney”, “Savannah”]:

... print(“Clean your room,”, kids, “!”)

...

Clean your room, Caleb !

Clean your room, Sydney !

Clean your room, Savannah !
```

Python allows you to use spaces or tabs. Python 2 was more permissive about mixed tabs and spaces in some cases, but Python 3 raises a TabError when indentation is inconsistent or ambiguous; even when code runs, mixing them makes troubleshooting harder. The standard for Python from the PEP 8 style guide is to use four spaces of indentation before each block of code. Why four spaces? Won’t one space work? Yes, it will, but your code blocks will be hard to align, and you will end up having to do extra work.

The alignment issue becomes especially important when you nest loops and conditional statements, as each loop needs to correspond to another block of code, indented using spaces. Many text editors allow you to view whitespace, and some even give you a visual indication of what is in a code block. Figure 3-2 shows this in VS Code.

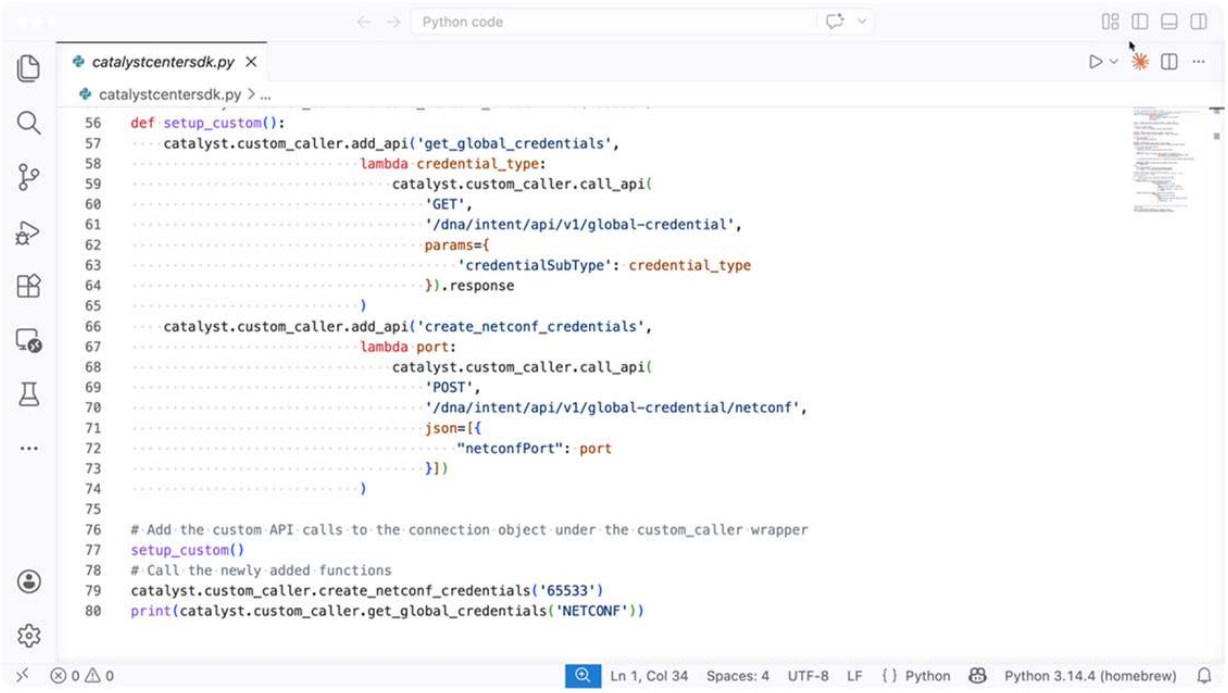


Figure 3-2 *Spaces and Code Blocks in Visual Studio Code*

Comments in Python are created by entering # or a string of three quotation marks (either single or double quotation marks). One very important good practice when coding is to write a description of what is happening in code that is not obvious. You probably will not want to write a comment for a simple **print** statement, but describing the output of a nested function would be useful for anyone who needs to make additions to your code in the future or to remind yourself why you did what you did during that late night caffeine-fueled coding session. The # is used to comment out a single line so the Python interpreter ignores it. Here is an example:

```
#get input from user in numeric format
```

The triple quote method is used to comment multiple lines and can be helpful when you want to provide a bit more context than what can fit on a single line of text. Here is an example:

```
''' This is

        line 2

        and line 3'''
```

Data Types and Variables

Data and variables are like the fuel and the fuel tank for a program. You can insert various types of data into a variable, and Python supports many types natively. Python can also be expanded with modules to support even more variables. A variable is really just a label that maps to a Python object stored somewhere in memory. Without variables, your programs would not be able to easily identify these objects, and your code would be a mess of memory locations.

Variables

Assigning a variable in Python is very straightforward. Python auto types a variable, and you can reassign that same variable to another value of a different type in the future. (Try doing that in C!) You just need to

remember the rules for variable names:

- A variable name must start with a letter or the underscore character.
- A variable name cannot start with a number.
- A variable name can only consist of alphanumeric characters and underscores (A–Z, 0–9, and _).
- A variable name is case sensitive (so Value and value are two different variable names).

To assign a variable you just set the variable name equal to the value you want, as shown in these examples:

Pip = "cat"	Variable assigned to a string
Age = 9	Variable assigned to an integer
Chill = True	Variable assigned to a Boolean
Variable1 = Variable2	Variable assigned to another variable

Data Types

Everything in Python is an object, and depending on the type of object, there are certain characteristics that you must be aware of when trying to determine the correct action you can perform on them. In Python, whenever you create an object, you are assigning that object an ID that Python uses to recall what is being stored. This mechanism is used to point to the memory location of the object in question and allows you to perform actions on it, such as printing its value. When the object is created, it is assigned a type that does not change. This type is tied to the object and determines whether it is a string, an integer, or another class.

Within these types, you are allowed to either change the object (mutable) or are not allowed to change the object (immutable) after it has been created. This doesn't mean that variables are not able to be changed; it means that most of the basic data types are not able to be modified but need to be replaced (or *assigned*, in Python speak) with another value. You can't just add a character at the end of a string value, for example. You have to instead reassign the whole string if you want to change it. This mutable/immutable concept will make more sense as you interact with various data types in programs. Python treats these two types of objects differently, and each has nuances that you must work around as you build your Python programs. To make it simple, think of immutable objects as ones that you want to stay the same, such as constants. Mutable objects, on the other hand, are objects that you will be adding elements to and subtracting from on a regular basis.

Table 3-2 lists the most commonly used Python data types. The rest of this section covers them in more detail.

Table 3-2 *Python Data Types*

Name	Type	Mutable	Description
Integer	int	No	Whole numbers, such as 6, 600, and 1589
Boolean	bool	No	Comparison value, either True or False
String	str	No	Sequence of characters delimited by quotes, such as “Cisco”, ‘Piper’, and “2000”
List	list	Yes	Ordered sequence of objects, such as [10, “DNA”, 19.8]
Tuple	tup	No	Ordered sequence of immutable objects, such as (10, “DNA”, 19.8)
Dictionary	dict	Yes	Unordered key:value pairs, such as {“key1”:”value1”,”name”:”Pip”}
Set	set	Yes	Unordered collection of unique objects, such as {“a”,”b”}

Integers, Floating Point, and Complex Numbers

The integers and floating point numbers are the simplest of data types:

- **Integers:** Whole numbers without decimal points
- **Floating point:** Numbers with decimal points or exponents (such as 10e5, which indicates 10 to the fifth power)

Python can perform advanced calculations, so it is used heavily in data science, and many features are built into the language and can be easily added with modules. For our purposes, though, the basic building blocks will suffice. Python has a simple set of built-in operators for working with numbers, much like the operators on a regular calculator. Table 3-3 lists Python’s numeric operators.

Table 3-3 *Python’s Numeric Operators*

Operator	Description	Example	Evaluates to
+	Adds two expressions together	5 + 5	10
-	Subtracts one expression from another	35 - 15	20
*	Multiplies two expressions	10 * 10	100
/	Divides one expression by another	20 / 5	4
//	Performs integer division (leaving off the remainder)	30 // 7	4
%	Performs modulus division (printing the remainder only)	30 % 7	2
**	Indicates an exponent	2 ** 8	256

When working with numbers in Python, a defined order of precedence must be observed in calculations. Python uses the following order (also known as PEMDAS):

1. **Parentheses:** Parentheses are always evaluated first.
2. **Power:** The exponent is evaluated.
3. **Multiplication:** Any multiplication is performed.
4. **Division:** Division is evaluated.
5. **Addition:** Addition is performed.
6. **Subtraction:** Subtraction is performed.
7. **Left to right:** After PEMDAS, anything else (such as **sqrt()** or other math functions) is evaluated from left to right.

In most languages, the parentheses are preferred over anything else. Most Python programmers use them liberally in their math formulas to make them simpler to construct without being so strict with the rules. Take the following example:

```
>>> 5 * 6 - 1

29
```

Python evaluates the multiplication first and then subtracts 1 from the result. If you wanted the subtraction to happen first, you could simply add parentheses around the parts you want evaluated:

```
>>> 5 * (6 - 1)

25
```

A floating point number is just a whole number with a decimal point. When you divide in Python, you often get a remainder that is displayed as a floating point number, as in this example:

```
>>> 10 / 7

1.4285714285714286
```

If you just want to see whole numbers, you can use integer division and lop off the remainder, as shown here:

```
>>> 10 // 7

1
```

Likewise, if you are only interested in the remainder, you can have modulus division show you what is left over:

```
>>> 10 % 7

3
```

You have the option to use other base systems instead of just the default base 10. You have three choices in addition to base 10: binary (base 2), octal (base 8), and hex (base 16). You need to use prefixes before integers in order for Python to understand that you are using a different base:

- 0b or 0B for binary
- 0o or 0O for octal
- 0x or 0X for hex

From Python's perspective, these are still just integers, and if you type any of them into the interpreter, it will return the decimal value by default, as shown in this example:

```
>>> 0xbadbeef

195935983
```

You can also convert back and forth by using the base keyword in front of the value you want to exchange, as in these examples:

```
>>> hex(195935983)

'0xbadbeef'



>>> bin(195935983)

'0b10111010110110111111011101111'
```

Booleans

A Boolean has only two possible values, **True** and **False**. You use comparison operators to evaluate between two Boolean objects in Python. This data type is the foundation for constructing conditional

steps and decisions within programs. Table 3-4 shows the various Boolean comparison operators and some examples of how to use them.

Table 3-4 *Boolean Comparisons*

Operator	What It Does	Example	Evaluates to
<	Less than	5 < 10	True
>	Greater than	6.5 > 3.5	True
<=	Less than or equal to	0 <= -5	False
>=	Greater than or equal to	6 >= 6	True
==	Equal to	5 = "5"	False
!=	Not equal to	5 != "5"	True

Strings

The string data type is a sequence of characters and uses quotes to determine which characters are included. The string **'Hello'** is just a set of characters that Python stores in order from left to right. Even if a string contains a series of numbers, it can still be a string data type. If you try to add a 1 to a string value, Python gives you an error, as shown in this example:

```
>>> '10' + 1

Traceback (most recent call last):

   File "<stdin>", line 1, in <module>

TypeError: can only concatenate str (not "int") to str
```

This error tells you that you have to convert the string to an integer or another data type to be able to use it as a number (in a math formula, for example). The **int()** function can convert a string value into an integer for you, as shown in this example:

```
>>> int('10') + 1

11
```

A string is just a list of characters in a certain order that Python keeps track of. In fact, this aspect of strings makes them easy to manipulate. If you use the string **'DevNet'**, you can pull out any individual characters of the string by knowing where it sits in the string index. One thing to keep in mind is that

indexes in Python always start with 0, so if you want the very first character in a string, your index value would be 0 and not 1. Figure 3-3 shows the string DevNet with its corresponding index values.

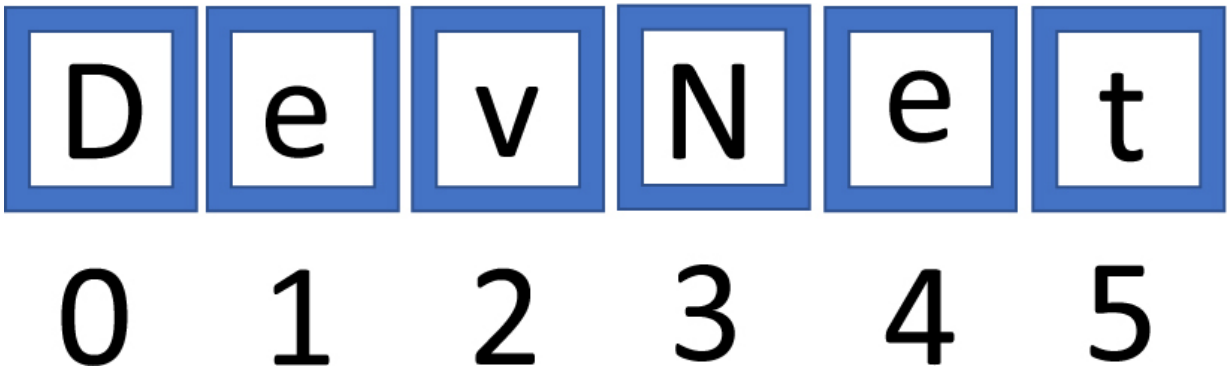


Figure 3-3 *DevNet String Index*

If you assign the string DevNet to a variable, you can separate and manipulate the component values of the string by using the index value. You can use brackets to specify the index number. The following example prints a capitol **D** from **DevNet**:

```
>>> a='DevNet'

>>> a[0]

'D'
```

You can also specify ranges to print. The colon operator gives you control over whole sections of a string. The first number is the beginning of the slice, and the second number determines the end. The second number may be confusing at first because it is intended to identify "up to but not including" the last character. Consider this example:

```
>>> a[0:3]

'Dev'
```

This example shows a **3** at the end, but this is technically four characters since the index starts at 0, but Python doesn't print the last character and instead stops right before it. For new Python programmers, this can be confusing, but remember that Python is literal. If you think of an index value as a box, in Figure 3-3, you want to stop at box 3 (but don't want to open the box). If you want to print the whole string, just pick a number beyond the index value, and Python will print everything, as in this example:

```
>>> a[0:6]

'DevNet'
```

If you omit a value for the first number, Python starts at 0, as in this example:

```
>>> a[:2]

'De'
```

If you omit the second value, Python prints to the end of the string, as in this example:

```
>>> a[2:]

'vNet'
```

You can also reverse direction by using negative numbers. If you put a negative first number, you start from the end of the string, as in this example:

```
>>> a[-2:]

'et'
```

A negative value on the other side of the colon causes Python to print using the end as a reference point, as in this example:

```
>>> a[:-2]

'DevN'
```

You can perform math operations on strings as well. The + is used to add or concatenate two strings together, as in this example:

```
>>> 'DevNet' + 'Rocks'

'DevNetRocks'
```

Multiplication works as well, as in this example:

```
>>> 'DevNet Rocks ' * 5

'DevNet Rocks DevNet Rocks DevNet Rocks DevNet Rocks DevNet Rocks '
```

There is a tremendous amount of string manipulation you can do with Python, and there are a number of built-in methods in the standard string library. These methods are called with a dot after the variable name for a string. Table 3-5 lists some commonly used string manipulation methods, including the syntax used for each and what each method does.

Table 3-5 *String Methods*

Method	What It Does
str.capitalize()	Capitalize the string
str.center(width[, fillchar])	Center justify the string
str.endwith(suffix[, start[, end]])	Add an ending string to the string
str.find(sub[, start[, end]])	Find the index position of the characters in a string
str.lstrip([chars])	Remove whitespace characters from the end of the string
str.replace(old, new[, count])	Replace characters in the string
str.lower()	Make the string all lowercase
str.rstrip([chars])	Strip whitespace characters from the front of the string
str.strip([chars])	Remove whitespace characters from the beginning and end of the string
str.upper()	Make the string all uppercase

Lists

Python, unlike other programming languages, such as C++ and Java, doesn't have arrays. If you want to store a bunch of values, you can use a list. You can use a variable to store a collection of items in a list. To create a list, you assign the contents of the list to a variable with the **=** and **[]** and separate the items with commas, as in this example:

```
>>> kids = ['Caleb', 'Sydney', 'Savannah']

>>> kids

['Caleb', 'Sydney', 'Savannah']
```

A list can contain any Python object, such as integers, strings, and even other lists. A list can also be empty and is often initialized in an empty state for programs that pull data from other sources. To initialize a list in an empty state, you just assign two brackets with nothing in them or you can use the built-in **list()** function:

```
emptylist = []

emptylist2 = list()
```

Lists are similar to strings in that each is a set of items indexed by Python that you can interact with and slice and dice. To pull out values, you just use the variable name with brackets and the index number, which starts at 0, as in this example:

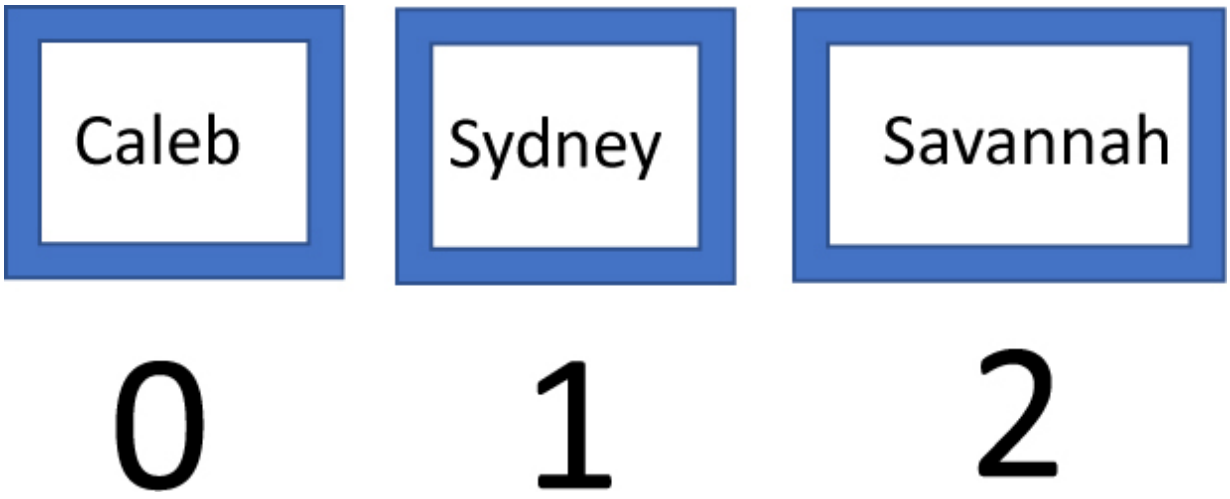


Figure 3-4 *List Index*

```
>>> print(kids[1])

Sydney
```

Figure 3-4 shows a list from the perspective of the index.

Unlike strings, lists are mutable objects, which means you can change parts of the list at will. With a string, you can't change parts of the string without creating a new string. This is not the case with lists, where you have a number of ways to make changes. If you have a misspelling, for example, you can change just one element of the list, leaving the rest untouched, as in this example:

```
>>> kids

['Caleb', 'Sidney', 'Savannah']

>>> kids[1]="Sydney"

>>> kids

['Caleb', 'Sydney', 'Savannah']

>>>
```

You can concatenate lists as well by using the + operator to join two lists together. The list items do not need to be unique. Python just connects the two together into a new list, as shown in the following example:

```
>>> a = [1, 2, 4]

>>> b = [4, 5, 6]

>>> c = a + b

>>> print(c)

[1, 2, 4, 4, 5, 6]
```

Remember all of the slicing you saw with strings? The same principles apply here, but instead of having a single string with each letter being in a bucket, the elements in the list are the items in the bucket. Don't forget the rule about the second number after the colon, which means "up to but not including." Here is an example:

```
>>> c= [1, 2, 3, 4, 5, 6, 7, 8, 9, 10]

>>> c[1:4]

[2, 3, 4]

>>> c[ :-4]

[1, 2, 3, 4, 5, 6]

>>> c[:]

[1, 2, 3, 4, 5, 6, 7, 8, 9, 10]
```

Table 3-6 describes some of the most common list methods.

Table 3-6 *List Methods*

Method	What It Does
list.append(element)	Adds an element to the end of the list
list.clear()	Removes everything from the list
list.copy(alist)	Returns a copy of the list
list.count(element)	Shows the number of elements with the specified value
list.extend(alist)	Adds the elements of a list to the end of the current list
list.index()	Returns the index number of the first element with a specified value
list.insert(index, element)	Adds an element at a specified index value
list.pop(index)	Removes an element at a specific index position, or if no index position is provided, removes the last item from the list
list.remove()	Removes a list item with a specified value
list.reverse()	Reverses the list order
list.sort()	Sorts the list alphabetically and/or numerically

Tuples

Tuples and lists are very similar. The biggest difference between the two comes down to mutability. As discussed earlier, Python data types are either mutable or immutable. Lists are mutable, and tuples are immutable. So why would you need these two types if they are so similar? It all comes down to how Python accesses objects and data in memory. When you have a lot of changes occurring, a mutable data structure is preferred because you don't have to create a new object every time you need to store different values. When you have a value that is constant and referenced in multiple parts of a program, an immutable data type (such as a tuple), is more memory efficient and easier to debug. You don't want some other part of your program to make changes to a crucial piece of data stored in a mutable data type.

To create a tuple, you use parentheses instead of brackets. You can use the **type()** function to identify a Python data type you have created. Here is an example:

```
>>> person = (2026, 'Mike', 'CCNA')

>>> person

(2026, 'Mike', 'CCNA')

>>> type(person)

<class 'tuple'>
```

You access data in a tuple the same way as in a list—by using brackets and the index value of the item in the tuple that you want to return:

```
>>> person[0]

2026
```

What you can't do with a tuple is make an assignment to one of the values:

```
>>> person[0]=15

Traceback (most recent call last):

 File "<stdin>", line 1, in <module>

TypeError: 'tuple' object does not support item assignment
```

Tuples can, however, be used to assign a set of variables quickly:

```
>>> (a, b, c) = (12, 'Fred',18)

>>> c

18
```

Dictionaries

A dictionary provides another way of creating a collection of items. Why do you need another way of storing data when you already have lists and tuples? A list is an ordered set of items tracked by an index. What if you need to access data that is tied to a certain value, such as a person's name? This capability is exactly why you need dictionaries in Python. A dictionary saves a ton of effort by giving you a built-in system for storing data in a key:value pair. As when you use labels on files in a filing cabinet, you can assign data to a key and retrieve it by calling that key as if you are pulling a file from the cabinet. Dictionaries don't have any defined order; all you need is the key—and not some index number—to get access to your data. There are some rules regarding dictionaries.

- **Keys:** A dictionary's keys are limited to only using immutable values (int, float, bool, str, tuple, and so on). No, you can't use a list as a key, but you can use a tuple, which means you could use a tuple as a key (immutable) but you can't use a list as a key (mutable).
- **Values:** A value can be any Python object or any combination of objects.

To create a dictionary, you use braces and your key and value separated by a colon. You separate multiple items with commas. Here's an example:

```
>>> cabinet = { "scores":(98,76,95), "name":"Chris", "company":"Cisco"}

>>> type(cabinet)

<class 'dict'>
```

Instead of using an index, you use the key, as shown in this example:

```
>>> cabinet["scores"]

(98, 76, 95)

>>> cabinet["company"]

'Cisco'
```

To add more items to a dictionary, you can assign them with a new key. You can even add another dictionary to your existing dictionary, as shown in this example:

```
>>> cabinet["address"] = {"street":"123 Anywhere Dr", "city":"Franklin", "state":"TN"}

>>> cabinet["address"]

{'street': '123 Anywhere Dr', 'city': 'Franklin', 'state': 'TN'}
```

Sets

A set in Python consists of an unordered grouping of data and is defined by using the curly braces of a dictionary, without the key:value pairs. Sets are mutable, and you can add and remove items from the set. You can create a special case of sets called a frozen set that makes the set immutable. A frozen set is often used as the source of keys in a dictionary (which have to be immutable); it basically creates a template for the dictionary structure. If you are familiar with how sets work in mathematics, the various operations you can perform on mutable sets in Python will make logical sense. To define a set, do the following:

```
>>> numbs = {1, 2, 4, 5, 6, 8, 10}

>>> odds = {1, 3, 5, 7, 9}
```

To check that these are indeed sets, use the **type()** function:

```
>>> type(odds)

<class 'set'>
```

To join two sets (just as in a mathematical join), you can use the **|** operator to show a combined set with no duplicates:

```
>>> numbs | odds

{1, 2, 3, 4, 5, 6, 7, 8, 9, 10}
```

You can get an intersection of two sets and show what numbers are in both by using the **&** operator:

```
>>> numbs & odds

{1, 5}
```

There are many ways to evaluate sets, and the Python documentation can be used to explore them all. In addition, Python has library collections that give you even more options for ways to store data. Search the Python documentation for "collections" and take a look at ordered dictionaries, named tuples, and other variations that may suit your data collecting needs. For the purposes of the CCNAAUTO exam, though, you do not need to know data structures beyond the basic types discussed here.

Input and Output

Input and output pretty much define what a computer does for you. In Python, the **input()** function and the **print()** function are two essential components that allow you to create interactive applications. In this section you will learn how to leverage these powerful functions in your applications.

Getting Input from the User

Python has the **input()** function to get information from a user running your Python code. The user is asked a question, and the program waits until the user types a response. It really is as simple as that. The **input()** function takes the characters that are entered and automatically stores them as a string data type, regardless of what the user enters. Here is an example:

```
>>> inpt = input('Type your name: ')

Type your name: Chris Jackson

>>> inpt

'Chris Jackson'
```

You assign a variable (in this case, **inpt**) to the **input()** function with a text prompt so that the user knows what is expected. That variable now holds the string the user typed. What if you need to get an integer or a floating point number? Since Python stores every input as a string, you need to do a conversion on the data supplied. Here is an example:

```
>>> inpt = float(input('What is the Temperature in F: '))

What is the Temperature in F: 83.5

>>> inpt

83.5
```

Here you asked the user for the temperature in Fahrenheit, which can be expressed as a floating point number. To make sure the variable holds the correct type, you used the **input()** function inside the **float()** function to convert the string into a floating point number.

The Mighty print() Function

The **print()** function provides output that can be displayed in the user's terminal. Just like every other function in Python, it is called with parentheses. This function is usually the gateway to writing your first code in Python, as in this example:

```
>>> print('Hello World')

Hello World
```

Every line printed with the **print()** function includes a newline character (**\n**) at the end, which is a special character that tells the terminal to advance one line. If you use the **\n** sequence within the **print()** string, it is interpreted as a new line, as shown in this example:

```
>>> print('Hello\nWorld')

Hello

World
```

There are numerous codes like this that can control how text is displayed in a string and how Python interprets the output. Without them, you would not be able to use, for example, a backslash in your text output. Here are a few of the most common ones:

- **\\:** Backslash
- **\b:** Backspace
- **\' :** Single quote
- **\":** Double quote
- **\t:** Tab
- **\r:** Carriage return

You can add multiple arguments to the **print()** function by using commas between elements. This is very useful in creating meaningful text, and the **print()** function also handles concatenation of the different data types automatically. Consider this example:

```
>>> print('Numbers in set', 1, ':', numbs )

Numbers in set 1 : {1, 2, 4, 5, 6, 8, 10}
```

By default, the **print()** function uses a separator between elements. This is normally not an issue if you want spaces to appear between words or elements. In the previous example, you can see a space between the **1** and the **:** that just doesn't look good. You can fix it by changing the separator that the **print()** functions uses with the **sep="** attribute (using single quotes with nothing in between). Since you will be removing all automatic spacing, you have to compensate for this by adding spaces in your actual text if you need them. Remember that separators come between elements and don't add anything to the start or end of the **print()** function. Consider this example:

```
>>> print('Numbers in set ', 1, ': ', numbs, sep='' )

Numbers in set 1: {1, 2, 4, 5, 6, 8, 10}
```

One capability added to Python 3.6 and up is the addition of f-string formatting. Not only are these strings easier to read and less prone to syntax errors but they allow you to write formatting code a lot faster. To create an f-string, you put an **f** at the beginning of a string, within the **print()** function, to let Python know what you are doing, and then you can use **{}** within your string to insert values or other functions. Here is an example:

```
>>> name = 'Piper'

>>> name2 = 'Chris'

>>> print(f'{name2} says Hi to {name}!')

Chris says Hi to Piper!
```

For more on formatting strings and beautifying your output, see the Python documentation.

Flow Control with Conditionals and Loops

So far you have been exposed to many of the building blocks of the Python language. The real power of a programming language is in the mechanisms you can use to embed logic and respond to different conditions by changing the flow of operation. Python has three primary control statements:

- **if:** An **if** statement is a conditional statement that can compare values and make branching decisions.
- **for:** A **for** loop is a counting loop that can iterate through data a specific number of times.
- **while:** The **while** loop can iterate forever when certain conditions are met.

You can use these three statements in various combinations to create very sophisticated programs. In this section you will see how each of these statements work.

If Statements

An **if** statement starts with an **if** and then sets up a comparison to determine the truth of the statement it is evaluating and ending with a **:** to tell Python to expect the clause (the action if the condition is true) block of code next. As mentioned earlier in this chapter, whitespace indenting matters very much in Python. The clause of an **if** statement must be indented (four spaces is the standard) from the beginning of the **if** statement. The following example looks for a condition where the variable **n** is equal to **5** and prints a message to the console indicating that the number is indeed a 5:

```
>>> n = 20

>>> if n == 20:

...     print('The number is 20')

...

The number is 20
```

The Python interpreter uses three dots to let you continue the clause for the **if** statement. Notice that there is space between the start of the dots and the **print()** statement. Without these four spaces, Python would spit back a syntax error like this:

```
>>> if n == 20:

... print(‘oops’)

  File “<stdin>”, line 2

    print(‘oops’)

 ^

IndentationError: expected an indented block
```

The goal of an **if** statement is to determine the “truth” of the elements under evaluation. This is Boolean logic, meaning that the operators evaluate True or False (refer to Table 3-4). The previous example is determining whether a variable is equal to a specific integer. What if the number is different? You might want to apply other logic by asking more questions. This is where else if (**elif** in Python) comes into play.

An **if** statement can have as many **elif** conditions as you want to add to the conditional check. Good coding practices recommend simplification, but there is no real limit to how many you add. Here is an example that uses two **elif** conditionals:

```
>>> n = 3

>>> if n == 17:

...     print('Number is 17')

... elif n < 10:

...     print('Number is less than 10')

... elif n > 10:

...     print('Number is greater than 10')

...

Number is less than 10
```

Since each **if** and **elif** statement does something only if the condition identified is true, it may be helpful to have a default condition that handles situations where none of the **if** or **elif** statements are true. For this purpose, you can assign a single **else** statement at the end, as shown in Example 3-2.

Example 3-2 *Adding a Final **else** Statement*

```
score = int(input('What was your test score?:'))

if score >= 90:
   print('Grade is A')
elif score >= 80:
   print('Grade is B')
elif score >= 70:
   print('Grade is C')
elif score >= 60:
   print('Grade is D')
else:
   print('Grade is F')

What was your test score?:53
Grade is F
>>>
```

For Loops

The **for** statement allows you to create a loop that continues to iterate through the code a specific number of times. It is also referred to as a counting loop and can work through a sequence of items, such as a list or other data objects. The **for** loop is heavily used to parse through data and is likely to be your go-to tool for working with data sets. A **for** loop starts with the **for** statement, followed by a variable name (which is a placeholder used to hold each sequence of data), the **in** keyword, some data set to iterate through, and then finally a closing colon, as shown in this example:

```
>>> dataset=(1,2,3,4,5)

>>> for variable in dataset:

...      print(variable)

...

1

2

3

4

5
```

The **for** loop continues through each item in the data set, and in this example, it prints each item. You can also use the **range()** function to iterate a specific number of times. The **range()** function can take arguments that let you choose what number it starts with or stops on and how it steps through each one. Here is an example:

```
>>> for x in range(3):

...      print(x)

...

0

1

2
```

By default, if you just give **range()** a number, it starts at 0 and goes by 1s until it reaches the number you provided. Zero is a valid iteration, but if you don't want it, you can start at 1. Consider this example:

```
>>> for x in range(1,3):

...     print(x)

...

1

2
```

To change the increment from the default of 1, you can add a third attribute to the **range()** statement. In the following example, you start at 1 and increment by 3 until you reach 10:

```
>>> for x in range(1,11,3):

...     print(x)

...

1

4

7

10
```

Remember that these ranges are up to and not including the final number you specify. If you want to go all the way to 10 in this example, you need to set your range to 11.

While Loops

Whereas the **for** loop counts through data, the **while** loop is a conditional loop, and the evaluation of the condition (as in **if** statements) being true is what determines how many times the loop executes. This difference is huge in that it means you can specify a loop that could conceivably go on forever, as long as the loop condition is still true. You can use **else** with a **while** loop. An **else** statement after a **while** loop executes when the condition for the **while** loop to continue is no longer met. Example 3-3 shows a **count** and an **else** statement.

Example 3-3 ***else*** *Statement with a* ***while*** *Loop*

```
>>> count = 1
>>> while (count < 5):
...     print("Loop count is:", count)
...     count = count + 1
... else:
...     print("loop is finished")
...
Loop count is: 1
Loop count is: 2
Loop count is: 3
Loop count is: 4
loop is finished
```

You can probably see similarities between the loop in Example 3-3 and a **for** loop. The difference is that the **for** loop was built to count, whereas this example uses an external variable to determine how many times the loop iterates. In order to build some logic into the **while** loop, you can use the **break** statement to exit the loop. Example 3-4 shows a **break** with **if** statements in an infinite **while** loop.

Example 3-4 *Using the **break** Statement to Exit a Loop*

```
while True:
    string = input('Enter some text to print. \nType "done" to quit> ')
    if string == 'done' :
       break
    print(string)
print('Done!')

Enter some text to print.
Type "done" to quit> Good luck on the test!
Good luck on the test!
Enter some text to print
Type "done" to quit> done
Done!
```

Notice the condition this example is checking with the **while** statement. Because **True** will always be **True** from an evaluation perspective, the **while** condition is automatically met. Without that **if** statement looking for the string **'done'**, your loop would keep asking for input and printing what you typed forever.

This chapter provides an overview of some of the key concepts and capabilities in Python. The goal is to prepare you to be able to read and understand code snippets that you might see on the CCNAAUTO exam. The next two chapters dive into other aspects of working with Python and Cisco APIs that are considered essential skills. Make sure you have followed along with the code examples here and that you are familiar with how to construct these basic examples. The following chapters build on these skills.

Type Hints

Python is dynamically typed, which means you don't have to declare what type a variable will be. This is great for quick scripts but can lead to surprising bugs and makes code harder to understand at a glance. Since Python 3.5, you can add type hints to your code that document the expected types for variables, function parameters, and return values. The Python interpreter ignores type hints at runtime (your code

still works the same way), but tools such as mypy, Pyright, and your IDE use them to catch errors before you run the code.

A typical typed function looks like this:

```
def add(x: int, y: int) -> int:

    return x + y
```

The colon after each parameter name introduces its type, and the arrow points to the return type. Variables can be typed similarly:

```
count: int = 0

name: str = "DevNet"

prices: list[float] = [9.99, 14.50, 24.00]
```

The standard library typing module provides building blocks for more complex types: Optional[str] for a value that might be None, dict[str, int] for a dictionary mapping strings to integers, Callable[[int], str] for a function that takes an int and returns a string, and so on. Modern Python (3.9+) lets you use the built-in collection types directly in hints (list[int] instead of List[int]).

You don't have to type every variable or function. Most teams add them incrementally, starting with the public APIs of their modules and working inward. The CCNAAUTO exam doesn't require you to write type hints, but you'll see them all over real-world Python code, so being able to read them is a practical skill.

AI Coding Assistants and the Exam

AI coding assistants (GitHub Copilot, Cursor, ChatGPT, Claude, Gemini) have become a normal part of modern software development. They can write a Python script that reads a CSV, parses some JSON, and posts to a REST API in seconds. That's genuinely useful and only going to get more so. But it raises a real question for anyone learning Python: if AI can write the code, why bother understanding it?

There are two answers, and both matter.

First, AI is excellent for studying. Cisco itself has invested heavily in this. New AI training and certifications such as the Cisco AI Technical Practitioner (AITECH) treat AI literacy as a core skill for technical professionals. When you're stuck on a syntax question, a confusing error message, or how a particular Python construct works, ask. AI assistants are excellent tutors for the kind of question you'd otherwise Google or post on Stack Overflow. They explain in plain language, give examples, and (usually) get the basics right. Use them as a personal study buddy: ask why something works, ask for an alternative way to do the same thing, ask what a piece of code in this book is doing.

Second, you will not have AI to help you on the CCNAAUTO exam. Cisco's exam policies prohibit any external assistance during a proctored exam, and the Pearson VUE testing environment doesn't allow access to outside tools, websites, notes, or applications. The exam tests what you know and can reason about, not what your AI assistant can produce. If you've leaned on AI to generate every example without understanding it, the exam will surface that gap fast.

That same caution applies well beyond the exam. AI-generated code you don't understand is a liability. You can't debug it when it breaks at three in the morning. You can't make sound architectural decisions. You can't tell when the AI confidently produces something subtly wrong (which it does, often). In the AI era, reading code well is more important than writing it by hand.

The practical takeaway: use AI to accelerate your learning, not to replace it. Read every line of code in this book and make sure you understand what it does. When you write your own scripts, use AI for quick scaffolding and for explaining unfamiliar libraries. But always read what you get back, change it to fit your situation, and verify it actually works. The fundamentals you build in this book are what let you tell good AI output from bad.

Exam Preparation Tasks

As mentioned in the section "How to Use This Book" in the Introduction, you have a couple of choices for exam preparation: the exercises here, Chapter 19, "Final Preparation," and the exam simulation questions on the companion website.

Review All Key Topics

Review the most important topics in this chapter, noted with the Key Topic icon in the outer margin of the page. Table 3-7 lists these key topics and the page number on which each is found.

Table 3-7 *Key Topics for Chapter 3*

Key Topic Element	Description	Page Number
Paragraph	Whitespace in Python code blocks	
Table 3-2	Python Data Types	
Paragraph	Strings	

Define Key Terms

There are no key terms for this chapter.

Additional Resources

Python Syntax: https://www.w3schools.com/python/python_syntax.asp

A Quick Tour of Python Language Syntax: https://jakevdp.github.io/WhirlwindTourOfPython/02-basic-python-syntax.html

PEP 8—Style Guide for Python Code: https://peps.python.org/pep-0008/

Coding & APIs: https://developer.cisco.com/startnow/#coding-apis-v0

Mutable vs Immutable Objects in Python: https://medium.com/@meghamohan/mutable-and-immutable-side-of-python-c2145cf72747

Your Guide to the Python print() Function: https://realpython.com/python-print/

Chapter 4. Python Functions, Classes, and Modules

This chapter covers the following topics:

- **Python Functions:** This section provides an overview of working with and building Python functions.
- **Using Arguments and Parameters:** This section describes how to use arguments and parameters to pass data into Python functions.
- **Object-Oriented Programming and Python:** This section describes key aspects of using object-oriented programming techniques.
- **Python Classes:** This section provides an overview of creating and using Python classes.
- **Working with Python Modules:** This section provides an overview of creating and using Python modules.
- **Type Hints in Functions and Classes:** This section shows how to use type hints to document what your functions and classes expect and return.

This chapter moves away from the basics introduced in Chapter 3, "Introduction to Python," and introduces Python functions, classes, and modules. Building Python functions allows for the creation of reusable code and is the first step toward writing object-oriented code. Classes are the Python tools used to construct Python objects and make it easier to produce scalable applications that are easy to maintain and readable. Finally, this chapter introduces the wide world of Python modules and how they can extend the capabilities of Python and make your job of coding much easier.

"Do I Know This Already?" Quiz

The "Do I Know This Already?" quiz allows you to assess whether you should read this entire chapter thoroughly or jump to the "Exam Preparation Tasks" section. If you are in doubt about your answers to these questions or your own assessment of your knowledge of the topics, read the entire chapter. Table 4-1 lists the major headings in this chapter and their corresponding "Do I Know This Already?" quiz questions. You can find the answers in Appendix A, "Answers to the 'Do I Know This Already?' Quiz Questions."

Table 4-1 *"Do I Know This Already?" Section-to-Question Mapping*

Foundation Topics Section	Questions
Python Functions	1–3
Using Arguments and Parameters	4
Object-Oriented Programming and Python	5–6
Python Classes	7–9
Working with Python Modules	10–11
Type Hints in Functions and Classes	12

Caution

The goal of self-assessment is to gauge your mastery of the topics in this chapter. If you do not know the answer to a question or are only partially sure of the answer, you should mark that question as wrong for purposes of self-assessment. Giving yourself credit for an answer that you correctly guess skews your self-assessment results and might provide you with a false sense of security.

1. Which of the following is the correct syntax for a Python function?

   a. define function (arg):

   b. function function(arg);

   c. def function(arg):

   d. func function(arg):

2. Which of the following is a valid Python function name?

   a. 1function

   b. __init__

   c. True

   d. Funct1on

3. When three single quotation marks are used on the next line directly after defining a function, what does this indicate?

   a. Multi-line text

   b. A docstring

   c. A string value including double or single quotation marks

   d. None of the above

4. Which of the following calls the function **sub()** using keyword arguments?

   a. sub(10, 15)

   b. sub(arg2=15, arg1=10)

   c. sub(*[10, 15])

d. sub(“arg1”, “arg2”)

5. What are key components of object-oriented programming in Python? (Choose two.)
   a. Functions that can be performed on a data structure
   b. Attributes that are stored in an object
   c. Configuration templates
   d. YAML files
6. Which of the following are benefits of OOP? (Choose all that apply.)
   a. Reusable code
   b. Easy to follow
   c. Low coupling/high cohesion
   d. Complex integration
7. Which of the following are used to define a class in Python? (Choose two.)
   a. class classname(parent):
   b. class classname:
   c. def class classname(arg):
   d. None of the above
8. What is a method?
   a. A variable applied to a class
   b. Syntax notation
   c. A function within a class or an object
   d. Something that is not used in a class
9. Which of the following describes inheritance?
   a. A hierarchy for functions in Python
   b. Class attributes and methods used as the starting point for another class
   c. A function only applied to methods being used in another class
   d. None of the above
10. Which module provides access to the file system and directory structure?
    a. filesystem
    b. open
    c. system
    d. os
11. Which module is a testing framework for Cisco infrastructure?
    a. pyATS
    b. pyang
    c. devnetats
    d. ncclient

12. What is the purpose of type hints in a Python function definition (for example, **def sub(arg1: int, arg2: int) -> int:**)?

    a. They document the expected argument and return types and enable editors and tools to catch type errors before the code runs.

    b. They cause the Python interpreter to reject a call that passes the wrong type at runtime.

    c. They automatically convert arguments to the annotated types.

    d. They are required in Python 3 for all function definitions.

Foundation Topics

Python Functions

In Python, a *function* is a named block of code that can take a wide variety of input parameters (or none at all) and return some form of output back to the code that called the function to begin with. It represents a key concept in programming sometimes referred to as DRY, which stands for Don't Repeat Yourself. The idea behind DRY is that if you perform some particular operations in your code multiple times, you can simply create a function to reuse that block of code anywhere you need it instead of duplicating effort by typing it each time.

Python offers two types of functions: built-in functions that are part of the standard library and functions you create yourself. The standard library includes a huge number of functions you can use in your program, like **print()**, many of which you have already been introduced to in Chapter 3. Building your own functions is how you construct capabilities that are not already present within the Python language.

To define a function in Python, you use the keyword **def**, a name for the function, a set of parentheses enclosing any arguments you want to pass to the function, and a colon at the end. The name of a function must follow these rules:

- Must not start with a number
- Must not be a reserved Python word, a built-in function (for example, **print()**, **input()**, **type()**), or a name that has already been used as a function or variable
- Can be any combination of the A–Z, a–z, 0–9 and the underscore (_)

The following is an example of an incredibly simple function that could be entered into the interactive Python interpreter:

```
Python 3.12.13 (main, Mar  3 2026, 09:30:42)

[Clang 17.0.0 (clang-1700.0.13.5)] on darwin

Type “help”, “copyright”, “credits” or “license” for more information.

>>> def devnet():

            ‘’’prints simple function’’’

             print(‘Simple function’)



>>> devnet()

Simple function
```

This function prints out the string “Simple function” any time you call it with **devnet()**. Notice the indented portion that begins on the next line after the colon. Python expects this indented portion to contain all the code that makes up the function. Keep in mind that whitespace matters in Python. The three single quotation marks that appear on the first line of the indented text of the function are called a *docstring* and can be used to describe what the function does.

As shown in the following example, you can use the built-in Python function **help()** to learn what a function does and any methods that can be used:

```
>>> help(devnet)

Help on function devnet in module __main__:



devnet()

       prints simple function
```

Using Arguments and Parameters

An argument is some value (or multiple values) that you pass to a function when you call the function within code. Arguments allow a function to produce different results and make code reuse possible. You simply place arguments within the parentheses after a function name. For example, this example shows how you can pass multiple numeric arguments to the **max()** function to have it return the largest number in the list:

```
>>> max(50, 5, 8, 10, 1)

50
```

Each function must define how it will use arguments, using parameters to identify what gets passed in and how it gets used. A parameter is simply a variable that is used in a function definition that allows code within the function to use the data passed to it. To get results back from the function, you use the keyword **return** and the object you want to pass back. The following example shows how to create a function that

subtracts two numbers and stores the result in a local variable called **result** that gets returned when the function is called:

```
>>> def sub(arg1, arg2):

          result = arg1 - arg2

          return result



>>> sub(10, 15)

-5
```

The variable **result** is local, meaning that it is not accessible to the main Python script, and it is used only within the function itself. If you tried to call **result** directly, Python would produce an error saying that **result** is not defined. You can, however, access global variables from within the function; you might do this, for example, to set certain constants or key variables that any function can use (for example, IP addresses). The difference in accessibility between a local variable and global variable is important, because they allow your code to maintain separation and can keep your functions self-contained.

The previous example uses positional arguments, which must be passed to a function in a particular order. Positional arguments work with a simple set of consistently applied arguments, but if a function needs more flexible alignment to parameters within a function, you can use keyword arguments instead. A *keyword argument* is a name/value pair that you pass to a function. Instead of using position, you specify the argument the function uses. It is a lot like assigning a variable to an argument. In the previous example, *arg1* is subtracted from *arg2*, and if the positions of these arguments were switched, you would get a different result when subtracting the values. With keyword arguments, it doesn't matter in what order they are passed to the function. Here is an example:

```
>>> sub(arg2=15, arg1=10)

-5
```

What happens if you don't know the total number of arguments that are being passed to a function? When you read in data, you might not know how many arguments to expect. Python allows you to use * and ** (often referred to as ***args** and ****kwargs**) to define any number of arguments or keyword arguments. * and ** allow you to iterate through a list or other collection of data, as shown in this example:

```
>>> def hello(*args):

         for arg in args:

             print("Hello", arg, "!")



>>> hello('Caleb', 'Sydney', 'Savannah')

Hello Caleb !

Hello Sydney !

Hello Savannah !
```

By using keyword arguments, you can send a list of key/value pairs to a function, as in the following example:

```
>>> def hello(**kwargs):

          for key, value in kwargs.items():

               print("Hello", value, "!")


>>> hello(kwarg1='Caleb', kwarg2='Sydney', kwarg3='Savannah')


Hello Caleb !

Hello Sydney !

Hello Savannah !
```

Note the use of the **items()** function in the **for** statement to unpack and iterate through the values.

You can also supply a default value argument in case you have an empty value to send to a function. By defining a function with an assigned key value, you can prevent an error. If the value in the function definition is not supplied, Python uses the default, and if it is supplied, Python uses what is supplied when the function is called and then ignores the default value. Consider this example:

```
>>> def greeting(name, message="Good morning!"):

          print("Hello", name + ', ' + message)


>>> greeting('Caleb')

Hello Caleb, Good morning!

>>> greeting('Sydney', "How are you?")

Hello Sydney, How are you?
```

Object-Oriented Programming and Python

Python was developed as a modern object-oriented programming (OOP) language. Object-oriented programming is a computer programming paradigm that makes it possible to describe real-world things and their relationships to each other. If you wanted to describe a router in the physical world, for example, you would list all its properties, such as ports, software versions, names, and IP addresses. In addition, you might list different capabilities or functions of the router that you would want to interact with. OOP was intended to model these types of relationships programmatically, allowing you to create an object that you can use anywhere in your code by just assigning it to a variable in order to instantiate it.

Objects are central to Python; in fact, Python really is just a collection of objects interacting with each other. An object is self-contained code or data, and the idea of OOP is to break up a program into smaller, easier-to-understand components. Up until now, you have mainly seen procedural programming techniques, which take a top-down approach and follow predefined sets of instructions. While this approach works well for simple programs, to write more sophisticated applications with better scalability, OOP is often the preferred method used by professional programmers. However, Python is very flexible in that you can mix and match these two approaches as you build applications.

Functions are an important part of the OOP principles of reusability and object-oriented structure. For the 200-901 CCNAAUTO exam, you need to be able to describe the benefits and techniques used in Python to build modular programs. Therefore, you need to know how to use Python classes and methods, which are covered next.

Python Classes

In Python, you use classes to describe objects. Think of a class as a tool you use to create your own data structures that contain information about something; you can then use functions (methods) to perform operations on the data you describe. A class models how something should be defined and represents an idea or a blueprint for creating objects in Python.

Creating a Class

Say that you want to create a class to describe a router. The first thing you have to do is define it. In Python, you define a class by using the **class** keyword, giving the class a name, and then closing with a colon. Pep8 (introduced in Chapter 3) recommends capitalizing a class name to differentiate it from a variable. Here is a simple example of creating a class in Python:

```
>>> class Router:

          pass
```

This example uses **pass** as a sort of placeholder that allows the class to be defined and set up to be used as an object. To make the class more useful, you can add some attributes to it. In the case of a router, you typically have some values that you want to have when you instantiate the class. Every router has a model name, a software version, and an IP address for management. You also need to pass some values to get started. The first value is always **self**. The reason for this will become obvious when you instantiate the class: The **self** value passes the object name that you select to instantiate the class. In the following example, the object you will create is **rtr1**:

```
class Router:

      '''Router Class'''

      def __init__(self, model, swversion, ip_add):

           '''initialize values'''

           self.model = model

           self.swversion = swversion

           self.ip_add = ip_add


rtr1 = Router('iosV', '15.6.7', '10.10.10.1')
```

After defining the class, you add a docstring to document what the class is for and then you create a function that calls **__init__**, which is a special case that is used for the setup of the class. (In **__init__**, the double underscores are called *dunder* or *magic methods*.) Functions that are within the class are called *methods* and become actions that you can perform on the object you are creating. To store attributes, you map the name **self** and the values you pass to it become variables inside the object, which then store those values as attributes. The last bit of code instantiates the object itself. Up until now, you have been creating a template, and by assigning data to the variables within the class, you have been telling Python to build the object. Now you can access any of the stored attributes of the class by using dot notation, as shown here:

```
>>> rtr1.model

'iosV'
```

When you call **rtr1.model**, the interpreter displays the value assigned to the variable model within the object. You can also create more attributes that aren't defined during initialization, as shown in this example:

```
>>> rtr1.desc = 'virtual router'

>>> rtr1.desc

'virtual router'
```

This example shows how flexible objects are, but you typically want to define any attributes as part of a class to automate object creation instead of manually assigning values. When building a class, you can instantiate as many objects as you want by just providing a new variable and passing over some data. Here is another example of creating a second router object **rtr2**:

```
>>> rtr2= Router('isr4221', '16.9.5', '10.10.10.5')

>>> rtr2.model

'isr4221'
```

Methods

Attributes describe an object, and methods allow you to interact with an object. Methods are functions you define as part of a class. In the previous section, you created an object and applied some attributes to it. Example 4-1 shows how you can work with an object by using methods. A method that allows you to see the details hidden within an object without typing a bunch of commands over and over would be a useful method to add to a class. Building on the previous example, Example 4-1 adds a new function called **getdesc()** to format and print the key attributes of your router. Notice that you pass **self** to this function only, as **self** can access the attributes applied during initialization.

Example 4-1 *Router Class Example*

```
class Router:
   '''Router Class'''
   def __init__(self, model, swversion, ip_add):
      '''initialize values'''
      self.model = model
      self.swversion = swversion
      self.ip_add = ip_add

   def getdesc(self):
      '''return a formatted description of the router'''
      desc = f'Router Model           :{self.model}\n'\
            f'Software Version       :{self.swversion}\n'\
            f'Router Management Address:{self.ip_add}'
      return desc

rtr1 = Router('iosV', '15.6.7', '10.10.10.1')
rtr2 = Router('isr4221', '16.9.5', '10.10.10.5')

print('Rtr1\n', rtr1.getdesc(), '\n', sep='')
print('Rtr2\n', rtr2.getdesc(), sep='')
```

There are two routers instantiated in this example: **rtr1** and **rtr2**. Using the **print** function, you can call the **getdesc()** method to return formatted text about the object's attributes. The following output would be displayed:

```
Rtr1

Router Model              :iosV

Software Version          :15.6.7

Router Management Address:10.10.10.1


Rtr2

Router Model              :isr4221

Software Version          :16.9.5

Router Management Address:10.10.10.5
```

Inheritance

Inheritance in Python classes allows a child class to take on attributes and methods of another class. In the previous section, Example 4-1 creates a class for routers, but what about switches? If you look at the **Router** class, you see that all of the attributes apply to a switch as well, so why not reuse the code already written for a new **Switch** class? The only part of Example 4-1 that wouldn't work for a switch is the **getdesc()** method, which prints information about a router. When you use inheritance, you can replace methods and attributes that need to be different. To inherit in a class, you create the class as shown earlier in this chapter, but before the colon, you add parentheses that include the class from which you want to pull attributes and methods. It is important to note that the parent class must come before the child class in the Python code. Example 4-2 shows how this works, creating a second class named **Switch**, using the **Router** class as parent. In addition, it creates a different **getdesc()** method that prints text about a switch rather than about a router.

Example 4-2 ***Router*** *Class and* ***Switch*** *Class with Inheritance*

```
class Router:
   '''Router Class'''
   def __init__(self, model, swversion, ip_add):
      '''initialize values'''
      self.model = model
      self.swversion = swversion
      self.ip_add = ip_add

   def getdesc(self):
      '''return a formatted description of the router'''
      desc = (f'Router Model           :{self.model}\n'
             f'Software Version       :{self.swversion}\n'
             f'Router Management Address:{self.ip_add}')
      return desc

class Switch(Router):
   def getdesc(self):
       '''return a formatted description of the switch'''
       desc = (f'Switch Model           :{self.model}\n'
              f'Software Version       :{self.swversion}\n'
              f'Switch Management Address:{self.ip_add}')
       return desc

rtr1 = Router('iosV', '15.6.7', '10.10.10.1')
rtr2 = Router('isr4221', '16.9.5', '10.10.10.5')
sw1 = Switch('Cat9300', '16.9.5', '10.10.10.8')

print('Rtr1\n', rtr1.getdesc(), '\n', sep='')
print('Rtr2\n', rtr2.getdesc(), '\n', sep='')
print('Sw1\n', sw1.getdesc(), '\n', sep='')
```

You can add another variable named **sw1** and instantiate the **Switch** class just as you did the **Router** class, by passing in attributes. If you create another **print** statement using the newly created **sw1** object, you see the output shown in Example 4-3.

Example 4-3 *Code Results of Using Class Inheritance*

```
Rtr1
Router Model            :iosV
Software Version         :15.6.7
Router Management Address:10.10.10.1

Rtr2
Router Model            :isr4221
Software Version         :16.9.5
Router Management Address:10.10.10.5

Sw1
Switch Model            :Cat9300
Software Version         :16.9.5
Switch Management Address:10.10.10.8
```

To learn more about classes, methods, and inheritance, you can refer to the Python documentation. https://docs.python.org/3/tutorial/classes.html

Working with Python Modules

A central goal of OOP is to allow you to build modular software that breaks code up into smaller, easier-to-understand pieces. One big file with thousands of lines of code would be extremely difficult to maintain and work with. If you are going to break up your code into functions and classes, you can also separate that code into smaller chunks that hold key structures and classes and allow them to be physically moved into other files, called *modules*, that can be included in your main Python code with the **import** statement. Creating modular code provides the following benefits:

- **Easier readability/maintainability:** Code written in a modular fashion is inherently easier to read and follow. It's like chapters in a book providing groupings of similar concepts and topics. Even the best programmers struggle to understand line after line of code, and modularity makes maintaining and modifying code much easier.
- **Low coupling/high cohesion:** Modular code should be written in such a way that modules do not have interdependencies. Each module should be self-contained so that changes to one module do not affect other modules or code. In addition, a module should only include functions and capabilities related to what the module is supposed to do. When you spread your code around multiple modules, bouncing back and forth, it is really difficult to follow. This paradigm is called low coupling/high cohesion modular design.
- **Code reusability:** Modules allow for easy reusability of your code, which saves you time and makes it possible to share useful code.
- **Collaboration:** You often need to work with others as you build functional code for an organization. Being able to split up the work and have different people work on different modules speeds up the code-production process.

There are a few different ways you can use modules in Python. The first and easiest way is to use one of the many modules that are included in the Python standard library or install one of thousands of third-party modules by using **pip**. Much of the functionality you might need or think of has probably already been written, and using modules that are already available can save you a lot of time. Another way to use

modules is to build them in the Python language by simply writing some code in your editor, giving the file a name, and appending a .py extension. Using your own custom modules does add a bit of processing overhead to your application, as Python is an interpreted language and has to convert your text into machine-readable instructions on the fly. Finally, you can program a module in the C language, compile it, and then add its capabilities to your Python program. Compared to writing your own modules in Python, this method results in faster runtime for your code, but it is a lot more work. Many of the third-party modules and those included as part of the standard library in Python are built this way.

Importing a Module

All modules are accessed the same way in Python: by using the **import** command. Within a program—by convention at the very beginning of the code—you type **import** followed by the module name you want to use. The following example uses the **math** module from the standard library:

```
>>> import math

>>> dir(math)

['__doc__', '__file__', '__loader__', '__name__', '__package__', '__spec__', 'acos',
   'acosh', 'asin', 'asinh', 'atan', 'atan2', 'atanh', 'ceil', 'comb', 'copysign',
'cos', 'cosh',
   'degrees', 'dist', 'e', 'erf', 'erfc', 'exp', 'expm1', 'fabs', 'factorial', 'floor',
'fmod',
   'frexp', 'fsum', 'gamma', 'gcd', 'hypot', 'inf', 'isclose', 'isfinite', 'isinf',
'isnan', 'isqrt',
   'ldexp', 'lgamma', 'log', 'log10', 'log1p', 'log2', 'modf', 'nan', 'perm', 'pi',
'pow',
   'prod', 'radians', 'remainder', 'sin', 'sinh', 'sqrt', 'tan', 'tanh', 'tau',
'trunc']
```

After you import a module, you can use the **dir()** function to get a list of all the methods available as part of the module. The ones in the beginning with the __ are internal to Python and are not generally useful in your programs. All the others, however, are functions that are now available for your program to access. As shown in Example 4-4, you can use the **help()** function to get more details and read the documentation on the **math** module.

Example 4-4 ***math*** *Module Help*

```
>>> help(math)
Help on module math:

NAME
   math

MODULE REFERENCE
   https://docs.python.org/3/library/math.html

   The following documentation is automatically generated from the Python
   source files.  It may be incomplete, incorrect or include features that
   are considered implementation detail and may vary between Python
   implementations.  When in doubt, consult the module reference at the
   location listed above.

DESCRIPTION
   This module provides access to the mathematical functions
   defined by the C standard.

FUNCTIONS
   acos(x, /)
      Return the arc cosine (measured in radians) of x.

   acosh(x, /)
      Return the inverse hyperbolic cosine of x.

   asin(x, /)
      Return the arc sine (measured in radians) of x.
-Snip for brevity-
```

You can also use **help()** to look at the documentation on a specific function, as in this example:

```
>>> help(math.sqrt)

Help on built-in function sqrt in module math:



sqrt(x, /)

     Return the square root of x.
```

If you want to get a square root of a number, you can use the **sqrt()** method by calling **math.sqrt** and passing a value to it, as shown here:

```
>>> math.sqrt(15)

3.872983346207417
```

You have to type a module's name each time you want to use one of its capabilities. This isn't too painful if you're using a module with a short name, such as **math**, but if you use a module with a longer name,

such as the **calendar** module, you might wish you could shorten the module name. Python lets you do this by adding **as** and a short version of the module name to the end of the **import** command. For example, you can use this command to shorten the name of the **calendar** module to **cal**.

```
>>> import calendar as cal
```

Now you can use **cal** as an alias for **calendar** in your code, as shown in this example:

```
>>> print(cal.month(2026, 2, 2, 1))




   February 2026

Mo Tu We Th Fr Sa Su

                    1

 2  3  4  5  6  7  8

 9 10 11 12 13 14 15

16 17 18 19 20 21 22

23 24 25 26 27 28
```

Importing a whole module when you need only a specific method or function adds unneeded overhead. To help with this, Python allows you to import specific methods by using the **from** syntax. Here is an example of importing the **sqrt()** and **tan()** methods:

```
>>> from math import sqrt,tan

>>> sqrt(15)

3.872983346207417
```

As you can see here, you can import more than one method by separating the methods you want with commas.

Notice that you no longer have to use **math.sqrt** and can just call **sqrt()** as a function, since you imported only the module functions you needed. Less typing is always a nice side benefit.

The Python Standard Library

The Python standard library, which is automatically installed when you load Python, has an extensive range of prebuilt modules you can use in your applications. Many are built in C and can make life easier for programmers looking to solve common problems quickly. Throughout this book, you will see many of these modules used to interact programmatically with Cisco infrastructure. To get a complete list of the modules in the standard library, go to at https://docs.python.org/3/library/. This documentation lists the modules you can use and also describes how to use them.

Importing Your Own Modules

As discussed in this chapter, modules are Python files that save you time and make your code readable. To save the class example from earlier in this chapter as a module, you just need to save all of the code for defining the class and the attributes and functions as a separate file with the .py extension. You can import your own modules by using the same methods shown previously with standard library modules. By default, Python looks for a module in the same directory as the Python program you are importing into. If it doesn't find the file there, it looks through your operating system's **path** statements. To print out the paths your OS will search through, consider this example of importing the **sys** module and using the **sys.path** method:

```
>>> import sys

>>> sys.path

['', '/Users/chrijack/Documents',
     '/Library/Frameworks/Python.framework/Versions/3.8/lib/python38.zip',
     '/Library/Frameworks/Python.framework/Versions/3.8/lib/python3.8',
     '/Library/Frameworks/Python.framework/Versions/3.8/lib/python3.8/lib-dynload',
     '/Library/Frameworks/Python.framework/Versions/3.8/lib/python3.8/site-packages']
```

Depending on your OS (this output is from a Mac), the previous code might look different from what you see here, but it should still show you what Python sees, so it is useful if you are having trouble importing a module.

If you remove the class from the code shown in Example 4-2 and store it in a separate file named device.py, you can import the classes from your new module and end up with the following program, which is a lot more readable while still operating exactly the same:

```
from device import Router, Switch



rtr1 = Router('iosV', '15.6.7', '10.10.10.1')

rtr2 = Router('isr4221', '16.9.5', '10.10.10.5')

sw1 = Switch('Cat9300', '16.9.5', '10.10.10.8')



print('Rtr1\n', rtr1.getdesc(), '\n', sep='')

print('Rtr2\n', rtr2.getdesc(), '\n', sep='')



print('Sw1\n', sw1.getdesc(), '\n', sep='')
```

When you execute this program, you get the output shown in Example 4-5. If you compare these results with the results shown in Example 4-3, you see that they are exactly the same. Therefore, the device module is just Python code that is stored in another file but used in your program.

Example 4-5 *Code Results of device.py Import as a Module*

```
Rtr1
Router Model            :iosV
Software Version         :15.6.7
Router Management Address:10.10.10.1

Rtr2
Router Model            :isr4221
Software Version         :16.9.5
Router Management Address:10.10.10.5

Sw1
Switch Model            :Cat9300
Software Version         :16.9.5
Router Management Address:10.10.10.8
```

Useful Python Modules for Cisco Infrastructure

This chapter cannot cover every single module that you might find valuable when writing Python code to interact with Cisco infrastructure. As you become more familiar with Python, you will come to love and trust a wide range of standard library and third-party modules. The following list includes many that are widely used to automate network infrastructure. Many of these modules are used throughout this book, so you will be able to see them in action. The following list provides a description of each one, how to install it (if it is not part of the standard library), and the syntax to use in your Python **import** statement:

- General-purpose standard library modules:
  - **pprint:** The pretty print module is a more intelligent print function that makes it much easier to display text and data by, for example, aligning data for better readability. Use the following command to import this module:

```
from pprint import pprint
```

  - **sys:** This module allows you to interact with the Python interpreter and manipulate and view values. Use the following command to import this module:

```
import sys
```

  - **os:** This module gives you access to the underlying operating system environment and file system. It allows you to open files and interact with OS variables. Use the following command to import this module:

```
import os
```

  - **datetime:** This module allows you to create, format, and work with calendar dates and time. It also enables timestamps and other useful additions to logging and data. Use the following command to import this module:

```
import datetime
```

  - **time:** This module allows you to add time-based delays and clock capabilities to your Python apps. Use the following command to import this module:

```
import time
```

- Modules for working with data:
  - **xmltodict:** This module translates XML-formatted files into native Python dictionaries (key/value pairs) and back to XML, if needed. Use the following command to install this module:

```
pip install xmltodict
```

    Use the following command to import this module:

```
import xmltodict
```

  - **csv:** This is a standard library module for understanding CSV files. It is useful for exporting Excel spreadsheets into a format that you can then import into Python as a data source. It can, for example, read in a CSV file and use it as a Python list data type. Use the following command to import this module:

```
import csv
```

  - **json:** This is a standard library module for reading JSON-formatted data sources and easily converting them to dictionaries. Use the following command to import this module:

```
import json
```

  - **PyYAML:** This module converts YAML files to Python objects that can be converted to Python dictionaries or lists. Use the following command to install this module:

```
pip install PyYAML
```

    Use the following command to import this module:

```
import yaml
```

  - **pyang:** This isn't a typical module you import into a Python program. It's a utility written in Python that you can use to verify your YANG models, create YANG code, and transform YANG models into other data structures, such as XSD (XML Schema Definition). Use the following command to install this module:

```
pip install pyang
```

- **Tools for API interaction:**
  - **requests:** This is a full library to interact with HTTP services and used extensively to interact with REST APIs. Use the following command to install this module:

```
pip install requests
```

    Use the following command to import this module:

```
import requests
```

  - **ncclient:** This Python library helps with client-side scripting and application integration for the NETCONF protocol. Use the following command to install this module:

```
pip install ncclient
```

    Use the following command to import this module:

```
from ncclient import manager
```

- **netmiko:** This connection-handling library makes it easier to initiate SSH connections to network devices. This module is intended to help bridge the programmability gap between devices with APIs and those without APIs that still rely on command-line interfaces and commands. It relies on the paramiko module and works with multiple vendor platforms. Use the following command to install this module:

```
pip install netmiko
```

  Use the following command to import this module:

```
from netmiko import ConnectHandler
```

- **pysnmp:** This is a Python implementation of an SNMP engine for network management. It allows you to interact with older infrastructure components without APIs but that do support SNMP for management. Use the following command to install this module:

```
pip install pysnmp
```

  Use the following command to import this module:

```
import pysnmp
```

- **Automation tools:**
  - **napalm:** napalm (Network Automation and Programmability Abstraction Layer with Multivendor Support) is a Python module that provides functionality that works in a multivendor fashion. Use the following command to install this module:

```
pip install napalm
```

    Use the following command to import this module:

```
import napalm
```

  - **nornir:** This is an extendable, multithreaded framework with inventory management to work with large numbers of network devices. Use the following command to install this module:

```
pip install nornir
```

    Use the following command to import this module:

```
from nornir.core import InitNornir
```

- Testing tools:
  - **unittest:** This standard library testing module is used to test the functionality of Python code. It is often used for automated code testing and as part of test-driven development methodologies. Use the following command to import this module:

```
import unittest
```

  - **pyats:** This framework was a gift from Cisco to the development community. pyATS is the core test infrastructure, and the Genie library runs on top of it to add device parsing, abstraction, and reusable test components. The combination is a powerful framework for automated infrastructure-as-code testing. Use the following command to install this module:

```
pip install pyats (just installs the core framework, check documentation for more
options)
```

    Many parts of the **pyats** framework can be imported. Check the documentation on how to use it.

Chapter 5, "Working with Data in Python," places more focus on techniques and tools used to interact with data in Python. This will round out the key Python knowledge needed to follow along with the examples in the rest of the book.

Type Hints in Functions and Classes

Type hints (introduced in Chapter 3) become especially valuable when you're writing functions, methods, and classes. They document what your code expects, help your IDE warn you about mistakes, and make refactoring safer. Take the sub function and the Router class from earlier in this chapter. Here's how they look with type hints added.

The sub function with type hints:

```
>>> def sub(arg1: int, arg2: int) -> int:

        result = arg1 - arg2

        return result
```

The Router class from Example 4-1, typed:

```
class Router:

    '''Router Class'''

    def __init__(self, model: str, swversion: str, ip_add: str) -> None:

        self.model: str = model

        self.swversion: str = swversion

        self.ip_add: str = ip_add
```

For classes whose primary job is holding data, the @dataclass decorator (introduced in Python 3.7) generates __init__, __repr__, and equality methods automatically from the typed attributes. This drops a lot of boilerplate. The same Router class as a dataclass:

```
from dataclasses import dataclass



@dataclass

class Router:

    '''Router Class'''

    model: str

    swversion: str

    ip_add: str
```

The two definitions of Router above are equivalent in behavior, but the dataclass version is shorter and you don't have to write the __init__ yourself. You'll see @dataclass widely in modern Python code,

anywhere you'd previously have written a class just to hold values.

Exam Preparation Tasks

As mentioned in the section "How to Use This Book" in the Introduction, you have a couple of choices for exam preparation: the exercises here, Chapter 19, "Final Preparation," and the exam simulation questions on the companion website.

Review All Key Topics

Review the most important topics in this chapter, noted with the Key Topic icon in the outer margin of the page. Table 4-2 lists these key topics and the page number on which each is found.

Table 4-2 *Key Topics*

Key Topic Element	Description	Page Number
Paragraph	Defining functions	
Paragraph	The value of object-oriented programming	
Paragraph	Defining classes	
Paragraph	Inheritance	
Paragraph	Python modules	
Bulleted list	Common Python modules	

Define Key Terms

There are no key terms for this chapter.

Chapter 5. Working with Data in Python

This chapter covers the following topics:

- **File Input and Output:** This section shows how to work with test files in Python.
- **Parsing Data:** This section discusses how to parse data into native Python objects.
- **Error Handling in Python:** This section discusses how to use try-except-else-finally to work through errors in Python input.
- **Test-Driven Development:** This section discusses using software testing to validate functions.
- **Unit Testing:** This section discusses how to use the internal Python module unittest to automate Python code testing.

There are numerous ways to ingest data into a Python program. You can get input from the user, pull data from a website or an API, or read data from a file. The trick is being able to convert data from a data source into native Python structures and objects so that you can use it to automate your infrastructure. This chapter discusses a number of ways to use built-in and third-party modules to transform different types of data into Python dictionaries, lists, and other data collection objects. The rest of the book provides more detail on how to use these techniques to interact with Cisco infrastructure; this chapter provides a foundation for understanding how these data formats differ and how best to interact with them.

"Do I Know This Already?" Quiz

The "Do I Know This Already?" quiz allows you to assess whether you should read this entire chapter thoroughly or jump to the "Exam Preparation Tasks" section. If you are in doubt about your answers to these questions or your own assessment of your knowledge of the topics, read the entire chapter. Table 5-1 lists the major headings in this chapter and their corresponding "Do I Know This Already?" quiz questions. You can find the answers in Appendix A, "Answers to the 'Do I Know This Already?' Quiz Questions."

Table 5-1 *"Do I Know This Already?" Section-to-Question Mapping*

Foundation Topics Section	Questions
File Input and Output	1, 2
Parsing Data	3–6
Error Handling in Python	7, 8
Test-Driven Development	9
Unit Testing	10, 11

Caution

The goal of self-assessment is to gauge your mastery of the topics in this chapter. If you do not know the answer to a question or are only partially sure of the answer, you should mark that question as wrong for purposes of self-assessment. Giving yourself credit for an answer that you correctly guess skews your self-assessment results and might provide you with a false sense of security.

1. When parsing a text file, what determines the end of a line?

   a. Return code

   b. Nothing; Python sees it as one big string

   c. \n or EoF

   d. All of the above

2. What syntax would you use to open a text file to be written to?

   a. data = open("text.txt", "w")

   b. data = load("text.txt", "w")

   c. load("text.txt", "w")

   d. open("text.txt", "w")

3. Which of the following do you use to write to a CSV file in Python?

   a. with open("text.csv", "a") as filehandle:
   csv_writer = csv.write(filehandle)
   csv_writer.writerow(data)

   b. with open("text.csv", "a") as filehandle:
   csv_writer.writerow(data)

   c. with open("text.csv", "a") as filehandle:
   csv_writer = csv.writer(filehandle)
   csv_writer.writerow(data)

   d. with open("text.csv", "a") as filehandle:
   csv_writer = csv.writer(f)
   csv_writer.writerow(data)

4. Which module is imported to read XML data?

   a. xmlm

   b. xmltodict

   c. XMLParse

   d. None of the above

5. Which methods are used for converting a native JSON file to Python and then back to JSON?

   a. load() and **dump()**

   b. loads() and **dump()**

   c. loads() and **dumps()**

   d. load() and **dumps()**

6. What does YAML stand for?

   a. Yet Another Markup Language

   b. YAML Ain't Markup Language

   c. The name of its creator

   d. None of the above

7. What is the syntax for error handling in Python?

   a. try-except-else-finally

   b. raise ErrorMessage

   c. assertErrorValue

   d. All of the above

8. When does the **finally** block execute?

   a. After the **try** block is successful

   b. After the **except** block

   c. At the end of every **try** block

   d. When an error code stops the **else** block

9. Test-driven development requires that developers:

   a. Create a unit test for every bit of code they write

   b. Know how to use DevOps tools for automated testing

   c. Create a simple test that fails and then write code that allows the test to succeed

   d. Completely unnecessary in an Agile development shop

10. What is the difference between a unit test and an integration test? (Choose two.)

    a. An integration test is for validation of how different parts of the application work together.

    b. An integration test verifies that the application operates as expected.

    c. A unit test verifies API functionality.

    d. A unit test is most specific in scope and tests small bits of code.

11. Which class is inherited as part of a unit test?

    a. unittest.testcase

    b. unittest.TestCase

    c. unittest

    d. TestCase

Foundation Topics

File Input and Output

Pulling data from a file in Python is very straightforward. To extract data from a text file, you can use native Python capabilities. Binary files, on the other hand, need to be processed by some module or another external program before it is possible to extract the data from them. The vast majority of your needs will be addressed through text files, so that is what this section focuses on.

From Python's perspective, a text file can be thought of as a sequence of lines. Each line, as it is read into Python, ends with a newline character at the end (**n** for Python). There are just two functions that you need to know when working with a text file: **open()** and **close()**.

To open a file and read in its contents, you have to first tell Python the name of the file you want to work with. You do this by using the **open()** function and assigning the output to a Python object (the variable **readdata** in this case). The function returns a file handle, which Python uses to perform various operations on the file. The code looks as follows:

```
readdata = open("textfile.txt", "r")
```

The **open()** function requires two arguments: the name of the file as a string and the mode that you want to open the file. In the preceding example, it opens the file in read mode. There are numerous options you can use when you set mode, and you can combine them in some cases to fine-tune how you want Python to handle the file. The following are some of the options:

- **r:** Open for reading (default)
- **w:** Open for writing, truncating the file first
- **x:** Open for exclusive creation, failing if the file already exists
- **a:** Open for writing, appending to the end of the file if it exists
- **b:** Open in binary mode
- **t:** Open in text mode (default)
- **+:** Open for updating (reading and writing)

With the previous code, you now have a file handling object named **readdata**, and you can use methods to interact with the file methods. To print the contents of the file, you can use the following:

```
print(readdata.read())
```

```
Line one of a text file

Line two of a text file, just like line one, but the second one.

Third line of a text file.
```

When using the **open()** function, you have to remember to close the file when you are finished reading from it. If you don't, the file will stay open, and you might run into file lock issues with the operating system while the Python app is running. To close the file, you simply use the **close()** method on the **readdata** object:

```
readdata.close()
```

Keeping track of the state of the file lock and whether you opened and closed it can be a bit of a chore. Python provides another way you can use to more easily work with files as well as other Python objects. The **with** statement (also called a *context manager* in Python) uses the **open()** function but doesn't require direct assignment to a variable. It also has better exception handling and automatically closes the file for you when you have finished reading in or writing to the file. Here's an example:

```
with open("textfile.txt", "r") as data:

 print(data.read())
```

This is much simpler code, and you can use all of the same methods to interact with the files as before. To write to a file, you can use the same structure, but in this case, because you want to append some data to the file, you need to change how you open the file to allow for writing. In this example, you can use **"a+"** to allow reading and appending to the end of the file. Here is what the code would look like:

```
with open("textfile.txt", "a+") as data:

 data.write('\nFourth line added by Python')
```

Notice the newline in front of the text you are appending to the file. It appears here so that it isn't just tacked on at the very end of the text. Now you can read the file and see what you added:

```
with open ("textfile.txt", "r") as data:

 print(data.read())



Line one of a text file

Line two of a text file, just like line one, but the second one.

Third line of a text file.

Fourth line added by Python
```

Parsing Data

Imagine a world where all the data is in nice, neatly formatted cells, completely structured, and always consistent. Unfortunately, data is not so easily accessible, as there are a multitude of types, structures, and formats. This is why it is essential that you learn how to parse data in some of the more common forms within your Python programs.

Comma-Separated Values (CSV)

A CSV file is just a plaintext spreadsheet or database file. All of those spreadsheets or databases that you have with infrastructure information can be easily exported as CSV files so that you can use them as source data in Python. Each line in a CSV file represents a row, and commas are used to separate the individual data fields to make it easier to parse the data. Python has a built-in CSV module that you can import that understands the CSV format and simplifies your code. The following is an example of a typical CSV file (in this case named routerlist.csv):

```
"router1","192.168.10.1","Nashville"

"router2","192.168.20.1","Tampa"

"router3","192.168.30.1","San Jose"
```

This example shows a common asset list or device inventory, such as one that you might pull from a network management system or simply keep track of locally. To start working with this data, you have to import the CSV module, and then you need to create a **reader** object to read your CSV file into. You first have to read the file into a file handle, and then you run the CSV read function on it and pass the results on to a **reader** object. From there, you can begin to use the CSV data as you wish. You can create another variable and pass the **reader** object variable to the built-in **list()** function. Here is what the code would look like:

```
>>> import csv

>>> samplefile = open('routerlist.csv')

>>> samplereader = csv.reader(samplefile)

>>> sampledata = list(samplereader)

>>> sampledata

[['router1', '192.168.10.1', 'Nashville'], ['router2', '192.168.20.1', 'Tampa'],

['router3',

  '192.168.30.1', 'San Jose ']]
```

In this example, you now have a list of lists that includes each row of data. If you wanted to manipulate this data, you could because it's now in a native format for Python. Using list notation, you can extract individual pieces of information:

```
>>> sampledata[0]

['router1', '192.168.10.1', 'Nashville']

>>> sampledata[0][1]

'192.168.10.1'
```

Using **with**, you can iterate through the CSV data and display information in an easier way:

```
import csv



with open("routerlist.csv") as data:

     csv_list = csv.reader(data)

     for row in csv_list:

        device = row[0]

        location = row[2]

        ip = row[1]

        print(f"{device} is in {location.rstrip()} and has IP {ip}.")
```

Notice the **rstrip** function used to format the **location** variable? You use it because the last entry in your CSV file will have a whitespace character at the very end when it is read into Python because it is at the very end of the file. If you don't get rid of it (by using **rstrip**), your formatting will be off.

The output of this code is as follows:

```
router1 is in Nashville and has IP 192.168.10.1.

router2 is in Tampa and has IP 192.168.20.1.

router3 is in San Jose and has IP 192.168.30.1.
```

If you want to add a fourth device to the CSV file, you can follow a process very similar to what you did with text files. Example 5-1 shows how to add a little interaction from the command line to fill in the fields and create a Python list with details on the new router. Instead using of a **reader** object, this example uses a **writer** object to store the formatted CSV data and then write it to the file.

Example 5-1 *Code and Input for a CSV File*

```
import csv

print("Please add a new router to the list")
hostname = input("What is the hostname? ")
ip = input("What is the ip address? ")
location = input("What is the location? ")


router = [hostname, ip, location]

with open("routerlist.csv", "a") as data:
  csv_writer = csv.writer(data)
  csv_writer.writerow(router)

<Below is interactive from the terminal after running the above code>
Please add a new router to the list
What is the hostname? router4
What is the ip address? 192.168.40.1
What is the location? London
```

If you run the code shown in Example 5-1 and input details for router 4, now when you display the router list, you have the new router included as well:

```
router1 is in Nashville and has IP 192.168.10.1.

router2 is in Tampa and has IP 192.168.20.1.

router3 is in San Jose and has IP 192.168.30.1.

router4 is in London and has IP 192.168.40.1.
```

JavaScript Object Notation (JSON)

JavaScript Object Notation (JSON) is a data structure that is derived from the JavaScript programming language, but it can be used as a portable data structure for any programming language. It was built to be an easily readable and standard way for transporting data back and forth between applications. JSON is heavily used in web services and is one of the core data formats you need to know how to use in order to interact with Cisco infrastructure. The data structure is built around key/value pairs that simplify mapping of data and its retrieval. Example 5-2 shows an example of JSON.

Example 5-2 *JSON*

```
{
    "interface": {
        "name": "GigabitEthernet1",
        "description": "Router Uplink",
        "enabled": true,
        "ipv4": {
            "address": [
                {
                    "ip": "192.168.1.1",
                    "netmask": "255.255.255.0"
                }
            ]
        }
    }
}
```

In Example 5-2, you can see the structure that JSON provides. **interface** is the main data object, and you can see that its value is multiple key/value pairs. This nesting capability allows you to structure very sophisticated data models. Notice how similar to a Python dictionary the data looks. You can easily convert JSON to lists (for a JSON array) and dictionaries (for JSON objects) with the built-in JSON module. There are four functions that you work with to perform the conversion of JSON data into Python objects and back.

- **load()**: This allows you to import native JSON and convert it to a Python dictionary from a file.
- **loads()**: This will import JSON data from a string for parsing and manipulating within your program.
- **dump()**: This is used to write JSON data from Python objects to a file.
- **dumps()**: This allows you to take JSON dictionary data and convert it into a serialized string for parsing and manipulating within Python.

The **s** at the end of **dump** and **load** refers to a string, as in *dump string*. To see this in action, you load the JSON file and map the file handle to a Python object (data) like so:

```
import json


with open("json_sample.json") as data:
    json_data = data.read()


json_dict = json.loads(json_data)
```

The object **json_dict** has taken the output of **json.loads(json_data)** and now holds the **json** object as a Python dictionary:

```
>>> type(json_dict)
<class 'dict'>


>>> print(json_dict)
{'interface': {'name': 'GigabitEthernet1', 'description': 'Router Uplink', 'enabled':
True,
    'ipv4': {'address': [{'ip': '192.168.0.2', 'netmask': '255.255.255.0'}]}}}
```

You can now modify any of the key/value pairs, as in this example, where the description is changed:

```
>>> json_dict["interface"]["description"] = "Backup Link"
>>> print(json_dict)
{'interface': {'name': 'GigabitEthernet1', 'description': 'Backup Link', 'enabled':
True,
    'ipv4': {'address': [{'ip': '192.168.0.2', 'netmask': '255.255.255.0'}]}}}
```

In order to save the new **json** object back to a file, you have to use the **dump()** function (without the **s**) to convert the Python dictionary back into a JSON file object. To make it easier to read, you can use the **indent** keyword:

```
with open("json_sample.json", "w") as fh:
    json.dump(json_dict, fh, indent = 4)
```

Now if you load the file again and print, you can see the stored changes, as shown in Example 5-3.

Example 5-3 *Loading the JSON File and Printing the Output to the Screen*

```
>>> with open ("json_sample.json") as data:
    json_data = data.read()
    print(json_data)
{
    "interface": {
        "name": "GigabitEthernet1",
        "description": "Backup Link",
        "enabled": true,
        "ipv4": {
            "address": [
                {
                    "ip": "192.168.0.2",
                    "netmask": "255.255.255.0"
                }
            ]
        }
    }
}
>>>
```

Extensible Markup Language (XML)

Extensible Markup Language (XML) is a very common data format that is used heavily in configuration automation. Parsing XML is similar to using other data formats, in that Python natively understands and can support XML encoding and decoding. The following is a very simple example of what XML structure looks like.

```
<device>

    <Hostname>Rtr01</Hostname>

    <IPv4>192.168.1.5</IPv4>

    <IPv6> </IPv6>

 </device>
```

It should come as no surprise that XML looks a bit like HTML syntax; both descend from SGML. XML is widely used for data transport and storage between web services and APIs. XML has a tree structure, with the root element being at the very top. There is a parent/child relationship between elements. In the preceding example, **device** is the root element that has **Hostname**, **IPv4**, and **IPv6** as child elements. Just like with HTML, a tag has meaning and is used to enclose the relationships of the elements with a start tag (<>)and a closing tag (</>). It's not all that different from JSON in that a tag acts as a key with a value. You can also assign attributes to a tag by using the following syntax:

```
attribute name="some value"
```

This works the same as an element in that it can provide a way to represent data. Example 5-4 shows an example of an IETF interface YANG model in XML.

Example 5-4 *YANG Model Represented in XML*

```
<?xml version="1.0" encoding="UTF-8" ?>
<interface xmlns="ietf-interfaces">
  <name>GigabitEthernet2</name>
  <description>Wide Area Network</description>
  <enabled>true</enabled>
  <ipv4>
    <address>
      <ip>192.168.1.5</ip>
      <netmask>255.255.255.0</netmask>
    </address>
  </ipv4>
</interface>
```

To work with this, you can use the native XML library, but it has a bit of a learning curve and can be a little hard to use if you just want to convert XML into something you can work with in Python. To make it easier, you can use a module called xmltodict to convert XML into an ordered dictionary in Python. This is a special class of dictionary that does not allow elements to change order. Since dictionaries use key/value pairs, where the key/value pairs are stored is normally not a problem, but in the case of XML, order matters. Example 5-5 reads in the XML from Example 5-4 and converts it to an ordered dictionary.

Example 5-5 *Reading in XML and Printing the Imported Dictionary to the Command Line*

```
import xmltodict

with open("xml_sample.xml") as data:
     xml_example = data.read()

xml_dict = xmltodict.parse(xml_example)


>>> print(xml_dict)
OrderedDict([('interface', OrderedDict([('@xmlns', 'ietf-interfaces'),
('name', 'GigabitEthernet2'), ('description', 'Wide Area Network'), ('enabled',
'true'), ('ipv4', OrderedDict([('address', OrderedDict([('ip', '192.168.0.2'),
('netmask', '255.255.255.0')]))]))]))])
```

Now that you have the XML in a Python dictionary, you can modify an element, as shown here:

```
xml_dict["interface"]["ipv4"]["address"]["ip"] = "192.168.55.3"
```

You can see your changes in XML format by using the **unparse** function (see Example 5-6). You can use the **pretty=True** argument to format the XML to make it a bit easier to read. XML doesn't care about whitespace, but humans do for readability.

Example 5-6 *Printing from the Command Line with the* ***unparse*** *Function*

```
>>> print(xmltodict.unparse(xml_dict, pretty=True))
<?xml version="1.0" encoding="utf-8"?>
<interface xmlns="ietf-interfaces">
     <name>GigabitEthernet2</name>
     <description>Wide Area Network</description>
     <enabled>true</enabled>
     <ipv4>
         <address>
              <ip>192.168.55.3</ip>
              <netmask>255.255.255.0</netmask>
         </address>
     </ipv4>
</interface>
```

To write these changes back to your original file, you can use the following code:

```
with open("xml_sample.xml", "w") as data:

 data.write(xmltodict.unparse(xml_dict, pretty=True))
```

YAML Ain't Markup Language (YAML)

YAML is an extremely popular human-readable format for constructing configuration files and storing data. It was built for the same use cases as XML but has a much simpler syntax and structure. It uses Python-like indentation to differentiate blocks of information and was actually built based on JSON syntax but with a whole host of features that are unavailable in JSON (such as comments). If you have ever used Docker or Kubernetes, you have undoubtably run into YAML files. The following is an example of YAML:

```
---

interface:

  name: GigabitEthernet2

  description: Wide Area Network

  enabled: true

  ipv4:

    address:

    - ip: 172.16.0.2

      netmask: 255.255.255.0
```

Notice that a YAML object has minimal syntax, all data that is related has the same indentation level, and key/value pairs are used to store data. YAML can also represent a list by using the **-** character to identify elements, as in the following example of IP addresses.

```
---

addresses:

  - ip: 172.16.0.2

    netmask: 255.255.255.0

  - ip: 172.16.0.3

    netmask: 255.255.255.0

  - ip: 172.16.0.4

    netmask: 255.255.255.0
```

To work with YAML in Python, you need to install and import the PyYAML module. Once you import it into your code, you can convert YAML to Python objects and back again. YAML objects are converted to dictionaries, and YAML lists automatically become Python lists. The two functions that perform this magic are **yaml.load** to convert from YAML objects into Python and **yaml.dump** to convert Python objects back to YAML. Just as in the other data examples, you can load a YAML file and then pass it to **yaml.load** to work its magic. The latest PyYaml module requires that you add an argument to tell it which loader you want to use. This is a security precaution so that your code will not be vulnerable to arbitrary code execution from a bad YAML file. Here is what the code looks like:

Caution

Always use yaml.safe_load() rather than yaml.load(). load() can execute arbitrary Python code embedded in a YAML file, which is a real security risk if the YAML comes from any untrusted source. The PyYAML maintainers explicitly recommend safe_load() for any YAML you don't fully control.

```
import yaml



with open("yaml_sample.yaml") as data:

    yaml_sample = data.read()



yaml_dict = yaml.safe_load(yaml_sample)
```

The variable **yaml_dict** is now a dictionary object containing your YAML file. Had this key/value been a YAML list, it would have created a list instead, like this:

```
>>> type(yaml_dict)

<class 'dict'>


>>> yaml_dict

{'interface': {'name': 'GigabitEthernet2', 'description': 'Wide Area Network',

'enabled':

    True, 'ipv4': {'address': [{'ip': '192.168.0.2', 'netmask': '255.255.255.0'}]}}}
```

As before, you can modify this object to your liking. For example, you can change the interface name to **GigabitEtherenet1**, as shown in Example 5-7.

Example 5-7 *Changing an Interface Name Within the Python Dictionary and Printing the Results*

```
>>> yaml_dict["interface"]["name"] = "GigabitEthernet1"

>>> print(yaml.dump(yaml_dict, default_flow_style=False))
interface:
  description: Wide Area Network
  enabled: true
  ipv4:
    address:
    - ip: 192.168.0.2
      netmask: 255.255.255.0
 name: GigabitEthernet1
```

To write these changes back to your file, use the following code.

```
with open("yaml_sample.yaml", "w") as data:

 data.write(yaml.dump(yaml_dict, default_flow_style=False))
```

> **Note**
>
> For more rigorous data validation when parsing JSON or YAML, the third-party pydantic library is widely used. You define a model class with typed fields, and pydantic validates incoming data against that model. It raises errors for missing fields, wrong types, or values outside allowed ranges. pydantic is the foundation of modern Python web frameworks like FastAPI and is worth learning once you're comfortable with the standard parsing libraries shown here.

Error Handling in Python

Whenever you are working with code, errors are bound to happen. In Python, errors often halt the execution of code, and the interpreter spits out some type of cryptic message. What if you wanted Python to tell the users what they did wrong and let them try again or perform some other task to recover from the error? That's where the **try-except-else-finally** code blocks come into play.

You have seen quite a bit of file access in this chapter. What happens if you ask the user for the filename instead of hard-coding it? If you did this, you would run the risk of a typo halting your program. In order to add some error handling to your code, you can use the **try** statement. Example 5-8 shows an example of how this works.

Example 5-8 ***try-except-else-finally*** *Code Example*

```
x = 0
while True:

    try:
        filename = input("Which file would you like to open? :")
        with open(filename, "r") as fh:
            file_data = fh.read()
    except FileNotFoundError:
        print(f'Sorry, {filename} doesn't exist! Please try again.')
    else:
        print(file_data)
        x = 0
        break
    finally:
        x += 1
        if x == 3:
            print('Wrong filename 3 times.\nCheck name and Rerun.')
           break
```

In this example, a variable keeps track of the number of times the **while** loop will be run. This is useful for building in some logic to make sure the program doesn't drive the users crazy by constantly asking them to enter a filename. Next is an infinite **while** loop that uses the fact that the Boolean True will always result in continuing looping through the code. Next is the **try** statement, which contains the block of code you want to subject to error handling. You ask the user to enter a filename to open, and it is stored in the **filename** variable. This variable is used with **open()** to open a read-only text file and use the file handle object **fh**. The file handle object uses **read()** to store the text file in the **file_data** variable. If Python can't find the file specified, the **except FileNotFoundError** block of code is executed, printing an error message with the file's name and informing the user to try again. The **else** block runs only if an exception does not occur and the filename can be found. The **file_data** is printed, **x** is set to **0** (to empty the counter), the loop is stopped, and the **finally** block is run. The **finally** block runs regardless of whether an exception occurs each time through the loop. The **x** variable is incremented each time through the loop, and if the user gets the wrong filename three times, a message is printed, saying the user tried three times and to check the file. At this point, the loop is broken, and the script is halted.

Here is what the program output would look like with a valid **test.txt** file in the script directory:

```
Which file would you like to open? :test

Sorry, test doesn't exist! Please try again.

Which file would you like to open? :test.txt

Test file with some text.

Two lines long.
```

Here is what the output would look like with three wrong choices:

```
Which file would you like to open? :test

Sorry, test doesn't exist! Please try again.

Which file would you like to open? :test2

Sorry, test2 doesn't exist! Please try again.

Which file would you like to open? :test3

Sorry, test3 doesn't exist! Please try again.

Wrong filename 3 times.

Check name and Rerun.
```

There are quite a few other error-handling capabilities available in Python, and if you want to try to make your applications more user friendly, it would be worth your time to explore them. The latest documentation can be found at https://docs.python.org/3/tutorial/errors.html. This documentation discusses custom errors and provides more examples of types of errors you can use with the previous sample code.

Test-Driven Development

Test-driven development (TDD) is an interesting concept that at first glance may seem completely backward. The idea is that you build a test case first, before any software has been created or modified. The goal is to streamline the development process by focusing on only making changes or adding code that satisfies the goal of the test. In normal testing, you test after the software is written, which means you spend your time chasing errors and bugs more than writing code. By writing the test first, you spend your time focused on writing only what is needed and making your code simple, easier to understand, and hopefully bug free. Figure 5-1 shows the TDD process in action.

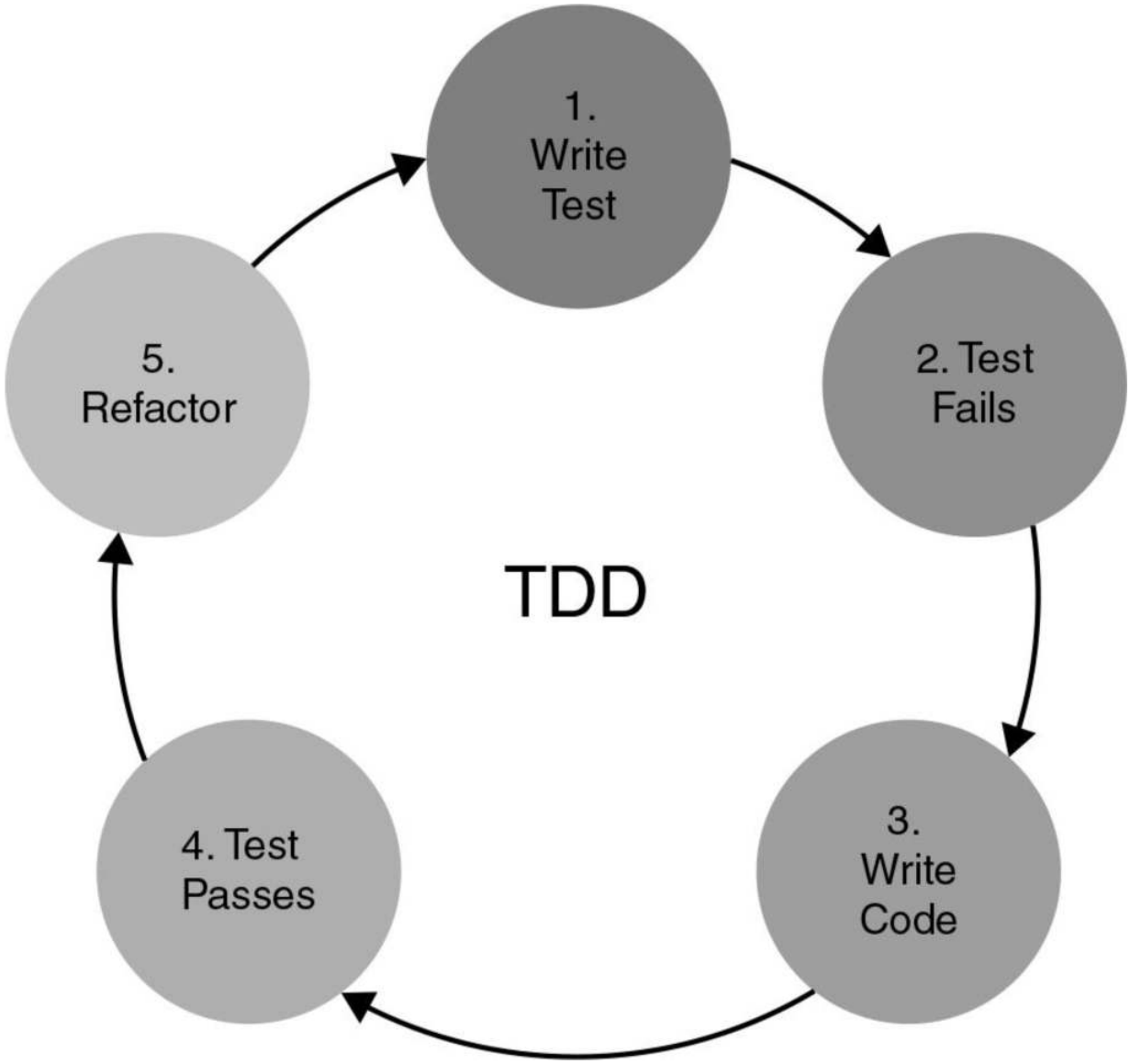


Figure 5-1 *Test-Driven Development in Action*

The following are the five steps of TDD:

Step 1. **Write a test:** Write a test that tests for the new class or function that you want to add to your code. Think about the class name and structure you will need in order to call the new capability that doesn't exist yet—and nothing more.

Step 2. **Test fails:** Of course, the test fails because you haven't written the part that works yet. The idea here is to think about the class or function you want and test for its intended output. This initial test failure shows you exactly where you should focus your code writing to get it to pass. This is like starting with your end state in mind, which is the most effective way to accomplish a goal.

Step 3. **Write some code:** Write only the code needed to make the new function or class successfully pass. This is about efficiency and focus.

Step 4. **Test passes:** The test now passes, and the code works.

Step 5. **Refactor:** Clean up the code as necessary, removing any test stubs or hard-coded variables used in testing. Refine the code, if needed, for speed.

TDD may see like a waste of time initially. Why write tests for stuff you know isn't going to pass? Isn't all of this testing just wasted effort? The benefit of this style of development is that it starts with the end goal in mind, by defining success right away. The test you create is laser focused on the application's purpose and a clear outcome. Many programmers add too much to their code by trying to anticipate future needs or building in too much complexity for an otherwise simple problem. TDD works extremely well with the iterative nature of Agile development, with the side benefit of having plenty of test cases to show that the software works.

Unit Testing

Testing your software is not optional. Every script and application that you create have to go through testing of some sort. Maybe it's just testing your syntax in the interactive interpreter or using an IDE and trying your code as you write it. While this is software testing, it's not structured and often is not repeatable. Did you test all options? Did you validate your expectations? What happens if you send unexpected input to a function? These are some of the reasons using a structured and automated testing methodology is crucial to creating resilient software.

A unit test is a type of test that is conducted on small, functional aspects of code. It's the lowest level of software testing and is interested in the logic and operation of only a single function in your code. That's not to say that you can't perform multiple tests at the same time. Computers are great at performing repetitive tasks, but the goal is for each test to be on one function at a time so that the testing is specific and consistent. Remember that a unit is the smallest testable part of your software.

There are other types of testing that you may hear about, such as integration testing and functional testing. The differences between these types of testing and unit testing come down to the scope of the test. As mentioned, a unit test is testing a small piece of code, such as a method or function. An integration test, on the other hand, tests how one software component works with the rest of the application. It is often used when modules of an application are developed by separate teams or when a distributed application has multiple components working together. A functional test (also called an end-to-end test) is the broadest in scope from a testing perspective. This is where the entire system is tested against the functional specifications and requirements of the software application. Figure 5-2 shows a testing pyramid to put it in perspective.

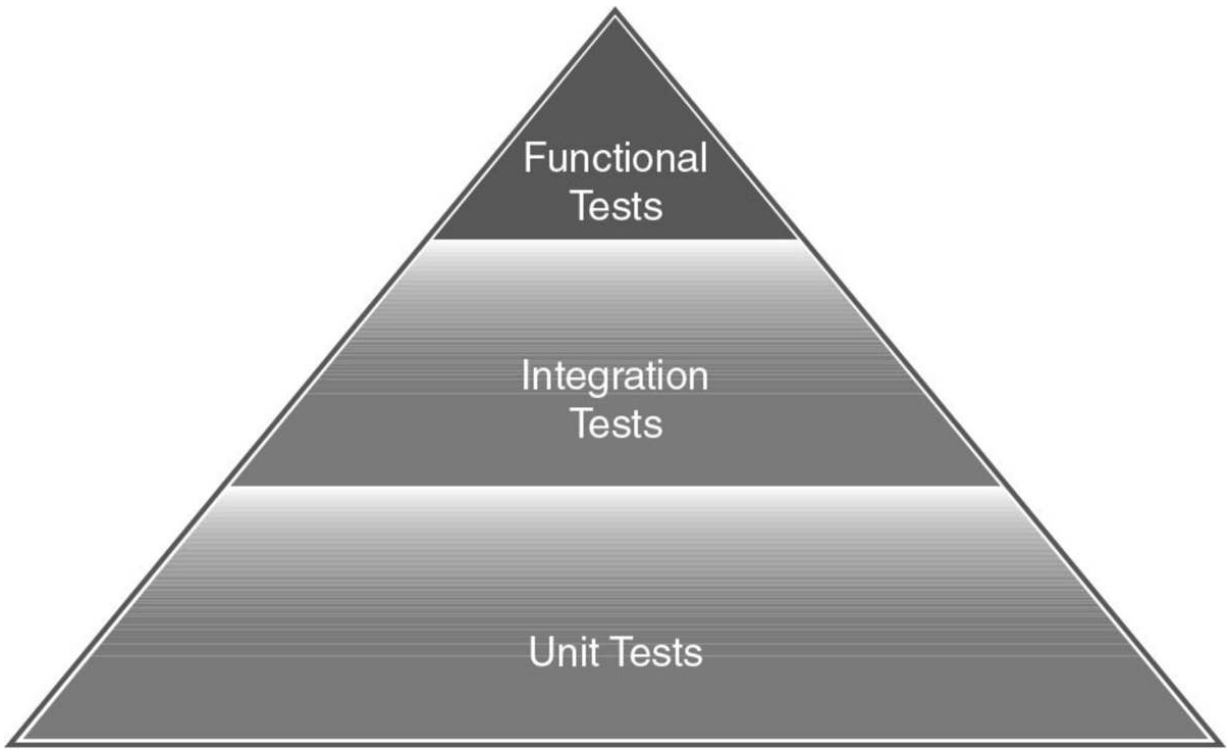


Figure 5-2 *Testing Pyramid*

Python has a built-in unit test module, named unittest. This module is quite full featured and can support a tremendous number of test cases. There are other testing modules out there (pytest is the most widely used), but for the purposes of the 200-901 CCNAAUTO exam, you need to know how unittest works. In order to use unittest, you need a bit of code to test. Here is a simple function that computes the area of a circle:

```
from math import pi



def area_of_circle(r):

    return pi*(r**2)
```

You import from the math module the pi constant to make it a little easier. A function is defined with the name **area_of_circle** that takes the argument **r**. The function computes the radius of a circle and returns the value. This is very simple, but what happens if the function is called, and odd values are passed to it? You guessed it: lots of errors. So, in order to test this function, you can create a unit test.

Certain conventions must be followed for a unit test. While you can create a unit test that has the code that you want to test all in the same file, it's a better idea to use object-oriented principles when building tests. In this case, the function you want to test is saved as areacircle.py, so following good practices you should name your unit test file test_areacircle.py. This makes it easy to differentiate the two. You should also import the unittest module, and from areacircle you can import the **area_of_circle** function. Import the **pi** method from math so that you can test your results. The **import** statements would look as follows:

```
import unittest

from areacircle import area_of_circle

from math import pi
```

Next, you need to create a class for your test. You can name it whatever you want, but you need to inherit **unittest.TestCase** from the unittest module. This is what enables the test function methods to be assigned to your **test** class. Next, you can define your first test function. In this case, you can test various inputs to validate that the math in your function under **test** is working as it should. You will notice a new method called **assertAlmostEqual()**, which takes the function you are testing, passes a value to it, and checks the returned value against an expected value. You can add a number of tests to this function. This is what the test now looks like with the additional code:

```
class Test_Area_of_Circle_input(unittest.TestCase):

   def test_area(self):

      # Test radius >= 0

     self.assertAlmostEqual(area_of_circle(1), pi)

     self.assertAlmostEqual(area_of_circle(0), 0)

     self.assertAlmostEqual(area_of_circle(3.5), pi * 3.5**2)
```

You can go to the directory where these two scripts reside and enter **python -m unittest test_areacircle.py** to run the test. If you don't want to type all that, you can add the following to the bottom of the **test_areacircle.py** script to allow the unittest module to be launched when you run the test script:

```
if __name__ == '__main__':

    unittest.main()
```

All this does is check to see if the script is being run directly (because the **__main__** special case is an attribute for all Python scripts run from the command line) and call the **unittest.main()** function. After executing the function, you should see the following results:

```
.

----------------------------------------------------------------------

Ran 1 test in 0.000s



OK
```

The dot at the top shows that 1 test ran (even though you had multiple checks in the same function) to determine whether the values submitted produced an error. Since all are valid for the function, the unit test came back successfully.

Now you can check to see if a negative number causes a problem. Create a new function under your previous **test_area** function. Name this function **test_values**. (The test at the beginning is required, or unittest will ignore the function and not check it.) You can use the **assertRaises** check, which will be looking for a **ValueError** exception for the function **area_of_circle**, and pass it a value of **-1**. The following function can be added to your code:

```
def test_values(self):

      # Test that bad values are caught

      self.assertRaises(ValueError, area_of_circle, -1)
```

Example 5-9 shows the output of the test with this additional check.

Example 5-9 *Output from Adding a New Test That Fails*

```
.F
======================================
FAIL: test_values (__main__.Test_Area_of_Circle_input)
----------------------------------------------------------------------
Traceback (most recent call last):
  File "/Users/chrijack/Documents/ccnadevnet/test_areacircle.py", line 14, in
  test_values
    self.assertRaises(ValueError, area_of_circle, -1)
AssertionError: ValueError not raised by area_of_circle

----------------------------------------------------------------------
Ran 2 tests in 0.001s

FAILED (failures=1)
```

The first check is still good, so you see one dot at the top, but next to it is a big **F** for fail. You get a message saying that the **test_value** function is where it failed, and see that your original function did not catch this error. This means that the code is giving bad results. A radius of **-1** is not possible, but the function gives you the following output:

```
>>> area_of_circle(-1)

3.141592653589793
```

To fix this, you go back to your original function and some error-checking code. You use a simple **if** statement to check for a negative number, and you raise a **ValueError** with a message to the user about invalid input:

```
from math import pi



def area_of_circle(r):

    if r < 0:

      raise ValueError('Negative radius value error')

     return pi*(r**2)
```

Now when you try the test from the interpreter, you see an error raised:

```
>>> area_of_circle(-1)

Traceback (most recent call last):

 File "<pyshell>", line 1, in <module>

 File "/Users/chrijack/Documents/ccnadevnet/areacircle.py", line 5, in area_of_circle

 raise ValueError('Negative radius value error')

ValueError: Negative radius value error
```

If you rerun the unit test, you see that it now passes the new check because an error is raised:

```
..

----------------------------------------------------------------------

Ran 2 tests in 0.000s


OK
```

This simple example barely scratches the surface of how you can use unit testing to check your software, but it does show you how a unit test is constructed and, more importantly, what it does to help you construct resilient code. Many more tests can be conducted; see the documentation at https://docs.python.org/3/library/unittest.html.

Real code often calls databases, APIs, or other systems that you don't want to hit during a unit test. The standard library's unittest.mock module lets you replace those dependencies with controlled stand-ins (mocks) that return whatever you specify. A test can verify your code's behavior in isolation without spinning up real infrastructure. Mocking is core to most non-trivial unit testing.

Coming back to pytest mentioned at the start of this section: it uses plain assert statements (no self.assertEqual boilerplate), supports parameterized tests and fixtures, and can run any unittest.TestCase class natively. Adopting pytest later doesn't mean rewriting your existing unittest code. Pick it up once the exam is behind you.

Exam Preparation Tasks

As mentioned in the section "How to Use This Book" in the Introduction, you have a couple of choices for exam preparation: the exercises here, Chapter 19, "Final Preparation," and the exam simulation questions on the companion website.

Review All Key Topics

Review the most important topics in this chapter, noted with the Key Topic icon in the outer margin of the page. Table 5-2 lists these key topics and the page number on which each is found.

Table 5-2 *Key Topics*

Key Topic Element	Description	Page Number
Section	JavaScript Object Notation (JSON)	
Section	Extensible Markup Language (XML)	
Section	Error Handling in Python	
Steps	Test-driven development	
Paragraph	Unit test in Python	
Example 5-9	Output from Adding a New Test That Fails	
Paragraph	Fixing a test that fails	

Define Key Terms

Define the following key terms from this chapter and check your answers in the glossary:

test-driven development (TDD)
unit test
integration test
functional test

Additional Resources

Reading Data from a File in Python: https://developer.cisco.com/learning/

Useful Python Libraries for Network Engineers: https://www.youtube.com/watch?v=Y4vfA11fPo0

Python unittest Module—How to Test Your Python Code? https://realpython.com/python-unittest/

Chapter 6. Application Programming Interfaces (APIs)

This chapter covers the following topics:

- **Application Programming Interfaces (APIs)**: This section describes what APIs are and what they are used for.
- **Representational State Transfer (REST) APIs**: This section provides a high-level overview of the RESTful APIs and how they function, as well as the benefits of using RESTful APIs.
- **RESTful API Authentication:** This section covers various aspects of the API authentication methods and the importance of API security.
- **Simple Object Access Protocol (SOAP)**: This section examines SOAP and typical examples of when and where this protocol is used.
- **Remote-Procedure Calls (RPCs):** This section provides a high-level overview of RPCs, why they are used, and the components involved.

Software developers use application programming interfaces (APIs) to communicate with and configure networks. APIs are used to communicate with applications and other software. They are also used to communicate with various network components through software. You can use APIs to configure or monitor specific network components, and there are multiple types of APIs. This chapter focuses on two of the most common APIs: northbound and southbound APIs. This chapter explains the differences between these types of APIs through the lens of network automation.

"Do I Know This Already?" Quiz

The "Do I Know This Already?" quiz allows you to assess whether you should read this entire chapter thoroughly or jump to the "Exam Preparation Tasks" section. If you are in doubt about your answers to these questions or your own assessment of your knowledge of the topics, read the entire chapter. Table 6-1 lists the major headings in this chapter and their corresponding "Do I Know This Already?" quiz questions. You can find the answers in Appendix A, "Answers to the 'Do I Know This Already?' Quiz Questions."

Table 6-1 *"Do I Know This Already?" Section-to-Question Mapping*

Foundation Topics Section	Questions
Application Programming Interfaces (APIs)	1, 2
Representational State Transfer (REST) APIs	3
RESTful API Authentication	4
Simple Object Access Protocol (SOAP)	5, 6
Remote-Procedure Calls (RPCs)	7

Caution

The goal of self-assessment is to gauge your mastery of the topics in this chapter. If you do not know the answer to a question or are only partially sure of the answer, you should mark that question as wrong for purposes of self-assessment. Giving yourself credit for an answer that you correctly guess skews your self-assessment results and might provide you with a false sense of security.

1. Which of the following is a sample use case of a southbound API?
   - **a.** Pushing network configuration changes down to devices
   - **b.** Increasing security
   - **c.** Streaming telemetry
   - **d.** Sending information to the cloud
2. What are some benefits of using asynchronous APIs? (Choose two.)
   - **a.** Not having to wait on a response to process data
   - **b.** Reduced processing time
   - **c.** Increased processing time
   - **d.** Data function reuse
3. What are the HTTP functions used for API communication? (Choose three.)
   - **a.** GET
   - **b.** SOURCE
   - **c.** PURGE
   - **d.** PATCH
   - **e.** PUT
4. True or false: RESTful API authentication can use API keys or custom tokens.

a. True

b. False

5. What does SOAP stand for?

a. Software Operations and Procedures

b. Software Operations Authentication Protocol

c. Simple Object Access Protocol

d. Simple Operations Automation Platform

e. Support Object Abstract Protocol

6. What are the main components of SOAP messages? (Choose all that apply.)

a. Envelope

b. Header

c. Destination

d. Body

e. Fault

f. Authentication

g. Source

7. Remote procedure calls (RPCs) behave similarly to which of the following?

a. Synchronous API

b. Asynchronous API

Foundation Topics

Application Programming Interfaces (APIs)

Software developers commonly use application programming interfaces (APIs) to communicate with and configure networks. APIs are mechanisms used to communicate with applications and other software. They are also used to communicate with various network components through software. APIs allow developers to programmatically configure or monitor specific network components, enabling automation and remote management. Although there are multiple types of APIs, this chapter focuses on two of the most common APIs: northbound and southbound. The following sections explain the differences between these two API types through the lens of network automation, and Figure 6-1 illustrates the typical basic operations of northbound and southbound APIs.

Northbound APIs

Northbound APIs are often used for communication from a network controller to its management software. For example, Cisco DNA Center has a graphical user interface (GUI) used to manage its network controller. Typically, when a network operator logs into a controller to manage the network, the information passed to the management software leverages a northbound REST-based API. Best practices suggest that the traffic should be encrypted using TLS between the software and the controller. Most APIs can use encryption to secure data in flight.

Note

RESTful APIs are covered in an upcoming section of this chapter and in-depth in Chapter 7, "RESTful API Requests and Responses."

Southbound APIs

If a network operator changes a switch's configuration in the controller's management software, those changes are then pushed down to the individual devices using a southbound API. These devices can be routers, switches, or even wireless access points. APIs interact with the components of a network through a programmatic interface. Southbound APIs can modify more than just the data plane on a device.

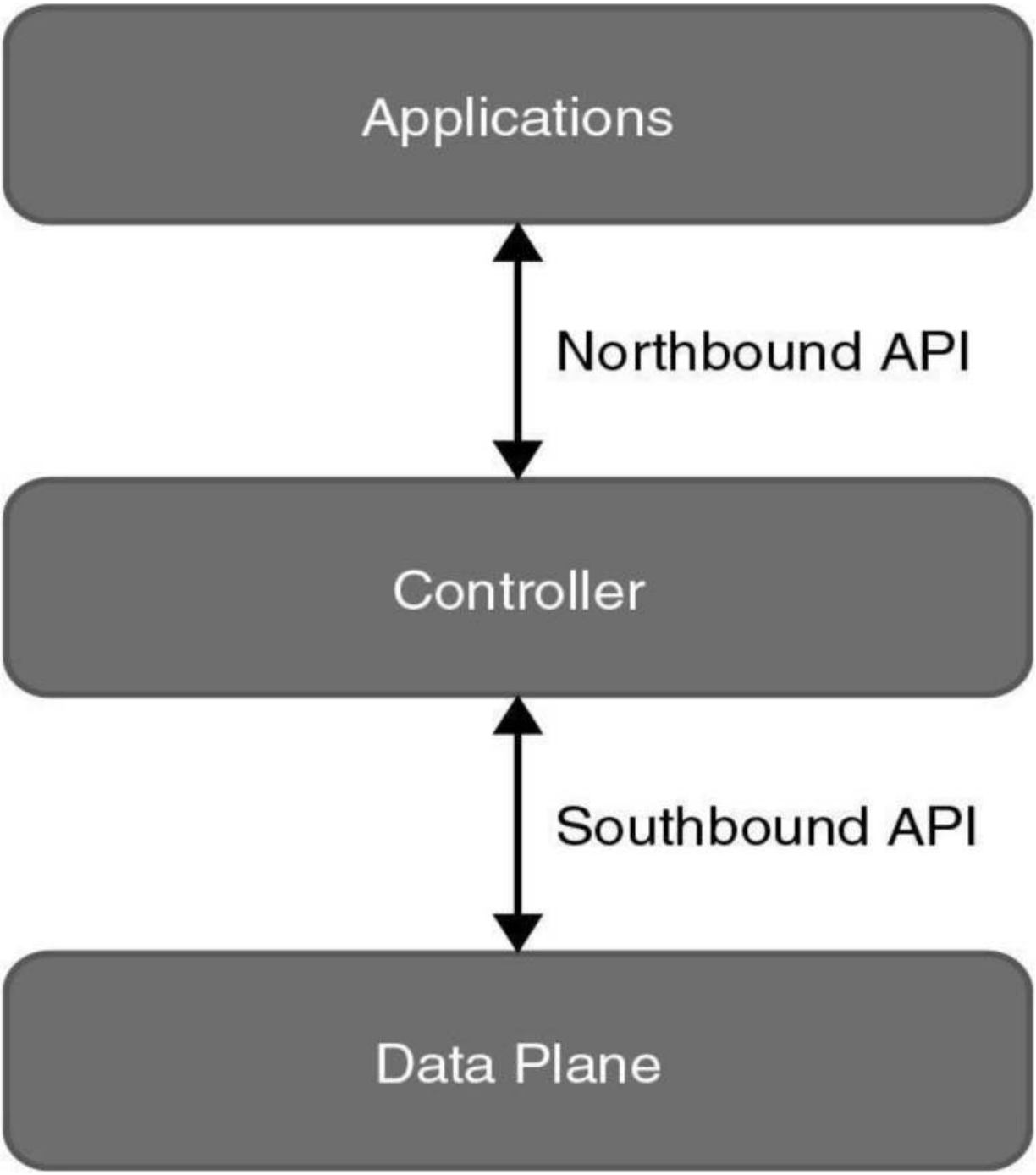


Figure 6-1 *Basic API Operations*

Synchronous Versus Asynchronous APIs

Application programming interfaces (APIs) can process transactions either synchronously or asynchronously. A synchronous API requires an application to wait for a response from the API to resume data processing or maintain normal functionality. This can interrupt application processing, as delays or failed responses may cause the application to stop functioning or behave in an unintended manner. This situation may arise, for instance, when an application relies on retrieving specific information from another API before it can proceed with its operations. To illustrate, uploading videos to YouTube was initially a synchronous use case. While the videos were being uploaded, users were unable to utilize the remainder of the graphical user interface (GUI) to modify video titles or make other alterations. Users were required to wait for the process to complete before engaging in any other activities within the YouTube application. Figure 6-2 illustrates an example of a synchronous process.

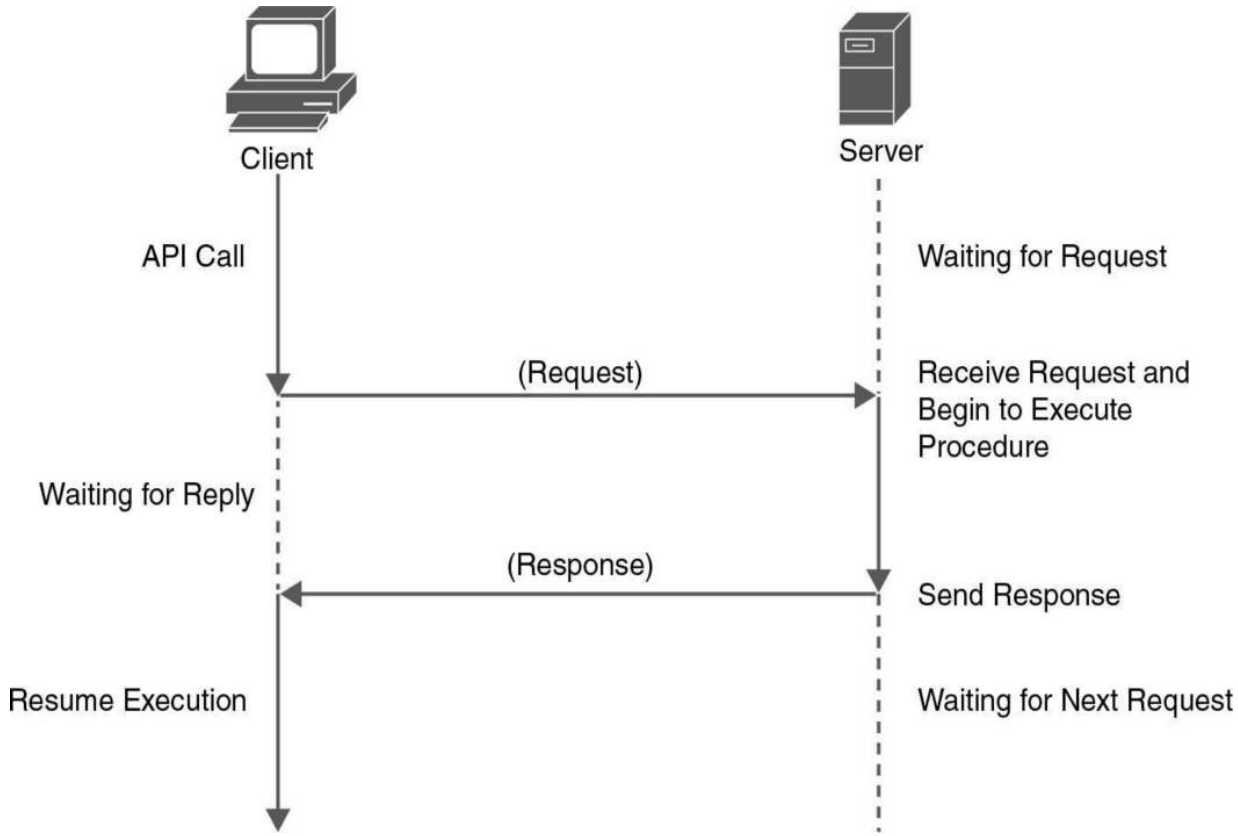


Figure 6-2 *Synchronous API Call Example*

Asynchronous APIs do precisely the opposite of synchronous APIs: they do not wait until a response is received before continuing to function and process other aspects of data. Asynchronous APIs provide a callback function so the API response can be sent back at another time without the application waiting for the entire transaction to complete. For example, you can upload a video to YouTube today, and while it's uploading, users can change the title, add hashtags, and even retrieve the URL to which the video will be posted once it is finished being uploaded.

The main difference between synchronous and asynchronous APIs lies in their execution flow: a synchronous API waits for other aspects of the code to complete before moving on and processing additional code. On the other hand, an asynchronous API lets the application continue processing while the API call completes, using a callback so it doesn't have to wait. An asynchronous API provides a better user experience, as users do not have to wait for specific information to be received before being able to use the application for other things. Figure 6-3 illustrates an asynchronous API call and how the application can continue processing while waiting for the response from the initial API call.

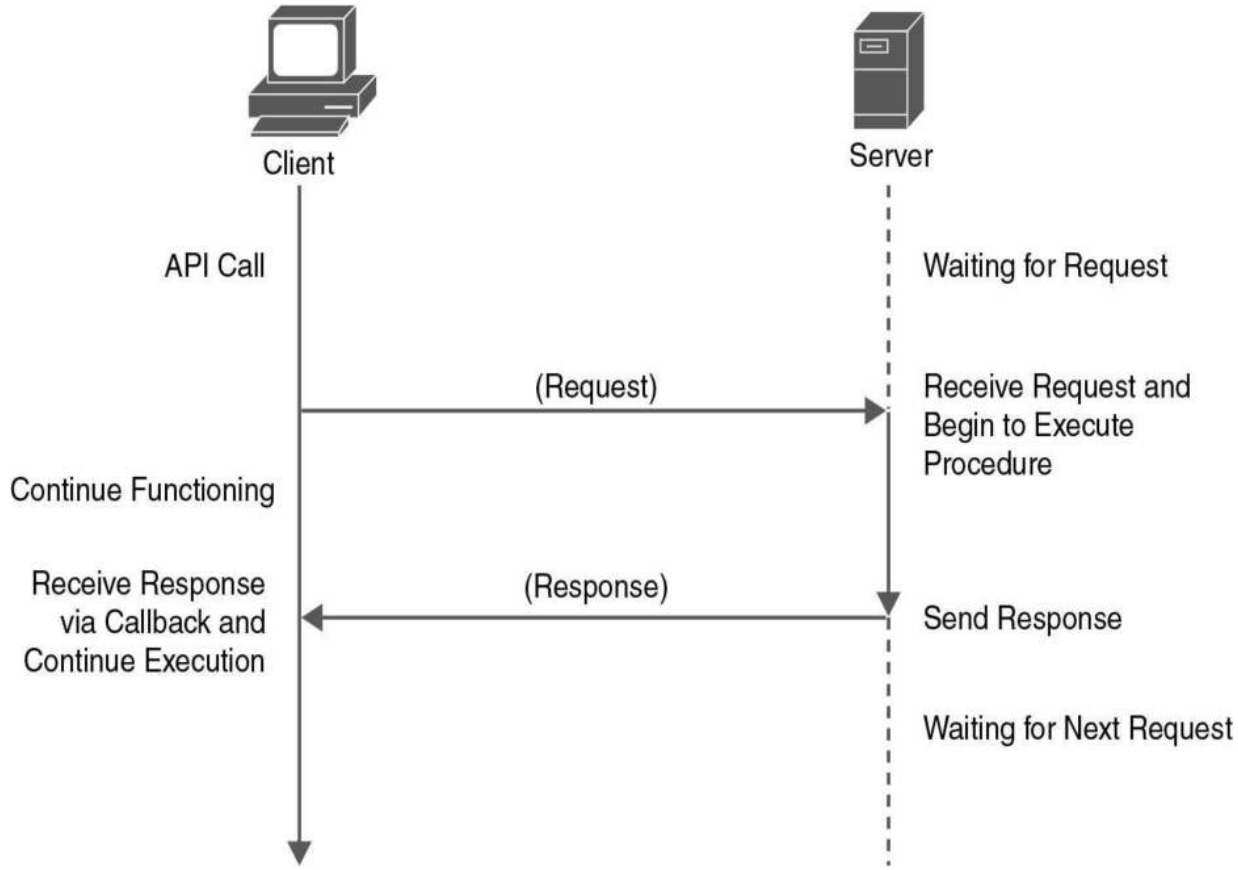


Figure 6-3 *Asynchronous API Call Example*

Representational State Transfer (REST) APIs

An API that uses REST is often referred to as a RESTful API. RESTful APIs use HTTP methods to gather and manipulate data. Because HTTP has a defined structure, it offers a consistent way to interact with APIs from multiple vendors. REST uses different HTTP functions to interact with data. Table 6-2 lists some of the most common HTTP functions and their associated use cases.

HTTP functions are very similar to the functions that most applications and databases use to store or alter data, whether stored in a database or within an application. These functions are called CRUD functions; CRUD is an acronym that stands for CREATE, READ, UPDATE, and DELETE. For example, in an SQL database, the CRUD functions are used to interact with or manipulate the data stored in the database. Table 6-3 lists the CRUD functions and their associated actions and use cases.

Table 6-2 *HTTP Functions and Sample Use Cases*

HTTP Function	Action	Use Case
GET	Requests data from a destination	Viewing a website
POST	Submits data to a specific destination	Submitting login credentials
PUT	Replaces data at a specific destination	Updating an NTP server
PATCH	Appends data to a specific destination	Adding an NTP server
DELETE	Removes data from a specific destination	Removing an NTP server

Table 6-3 *CRUD Functions and Sample Use Cases*

CRUD Function	Action	Use Case
CREATE	Inserts data inside a database or an application	Creating a customer's home address in a database
READ	Retrieves data from a database or an application	Pulling up a customer's home address from a database
UPDATE	Modifies or replaces data in a database or an application	Changing a street address stored in a database
DELETE	Removes data from a database or an application	Removing a customer from a database

Whether you are trying to learn how APIs interact with applications or controllers, test code and outcomes, or become a full-time developer, testing is one of the most important pieces of interacting with any software via APIs. Testing code helps ensure that developers are accomplishing the desired outcome. This chapter covers some tools and resources that make it possible to practice using APIs and REST functions and hone your development skills to become a more efficient network engineer with coding skills.

Note

Chapter 7 provides more detail on HTTP and CRUD functions and response codes.

RESTful API Authentication

As mentioned earlier in this chapter, it is essential to be able to interact with a software controller using RESTful APIs and to test code to see whether the desired outcomes are accomplished when executing the code. Remember that APIs are software interfaces into an application or a controller. Many APIs require authentication; such APIs are like devices in that the user must authenticate to access them. Once a user has authenticated, any changes a developer can make via the API can impact the application. This means

if a RESTful API call is used to delete data, that data will be removed from the application or controller just as if a user were logged into the device via the CLI and deleted it. It is best practice to use a test lab or a Cisco DevNet sandbox while learning or practicing API concepts to prevent accidental impacts in a production or lab environment.

> **Note**
>
> Cisco DevNet is covered in Chapter 1, "Introduction to Cisco DevNet Associate Certification."

Basic Authentication

Figure 6-4 illustrates basic authentication as one of the APIs' most straightforward and common authentication methods. The downfall of basic authentication is that the credentials are passed unencrypted. This means that if the transport is simple HTTP, sniffing the traffic and capturing the username and password with little to no effort is possible. The lack of encryption means the credentials are in simple plaintext Base64 encoding in the HTTP header. However, basic authentication is more commonly used with SSL or TLS to prevent such attacks.

Another big issue with basic authentication is that the password is sent back and forth with each request, which increases the opportunity for an attacker to capture the traffic containing the password. This is yet another reason to use encryption on this type of transaction.

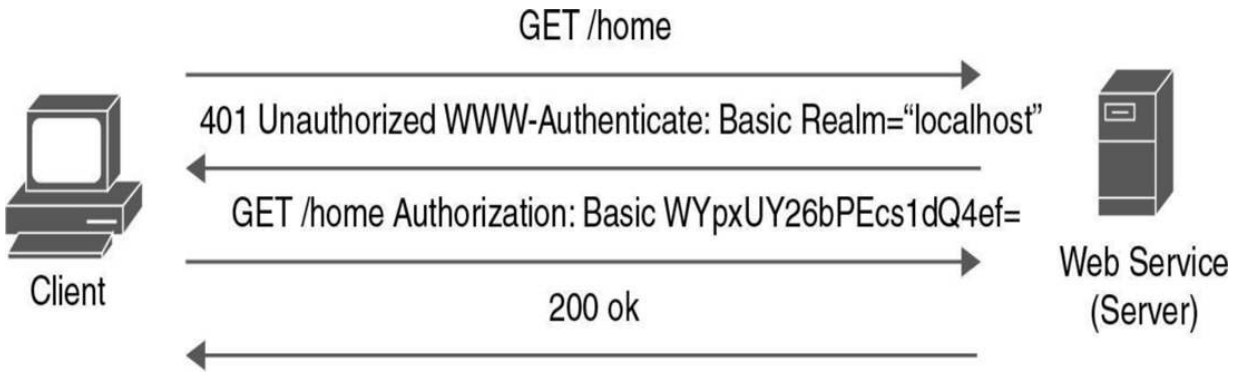


Figure 6-4 *Basic Authentication Example*

API Keys

Some APIs use API keys for authentication. An API key is a predetermined string passed from the client to the server. It is intended to be a pre-shared secret and should not be well-known or easy to guess because it functions like a password. Anyone with this key can access the API in question, which can potentially cause a significant outage and gain access to critical or sensitive data. An API key can be passed to the server in three different ways:

- String
- Request header
- Cookie

Example 6-1 provides an example of a string-based API key. This type of API key is sent with every API call and is often used as a one-off authentication method. When you're looking to do multiple API calls, it isn't convenient to manually enter the API key string every time. This is where request headers or cookies come into play.

Example 6-1 *String-Based API Key Example*

```
GET /something?api_key=abcdef12345
```

Request headers are frequently used when a user makes multiple API calls and doesn't want to keep putting the API key into each API individually. This approach is typically seen in Postman and Python scripts. The header must include the string or token in the header of each API call. Example 6-2 shows the request header option for API key authentication.

Example 6-2 *Request Header API Key Example*

```
GET /something HTTP/1.1
X-API-Key: abcdef12345
```

Finally, one of the most common methods for recurring API calls is to use cookies. A cookie stores the API key string and can be reused and stored repeatedly. This is synonymous with a header. Example 6-3 shows an API-key cookie that uses the same key as the previous examples.

Example 6-3 *Cookie API Key Example*

```
GET /something HTTP/1.1
Cookie: X-API-KEY=abcdef12345
```

> **Note**
>
> Later chapters provide detailed examples of the authentication methods introduced in this chapter.

Custom Tokens

A custom token allows a user to enter their username and password once and receive a unique, auto-generated, and encrypted token. The user can then use this token to access protected pages or resources instead of continuously entering the login credentials. Tokens can be time-bound and set to expire after a specific amount of time has passed, thus forcing users to reauthenticate by reentering their credentials. A token is designed to show proof that a user has previously authenticated. It simplifies the login process and reduces the number of times a user has to provide login credentials. A token is stored in the user's browser and gets checked each time the user tries to access information requiring authentication. Once the user logs out of the web browser or website, the token is destroyed so it cannot be compromised. Figure 6-5 provides an overview of token-based authentication between a client and a server.

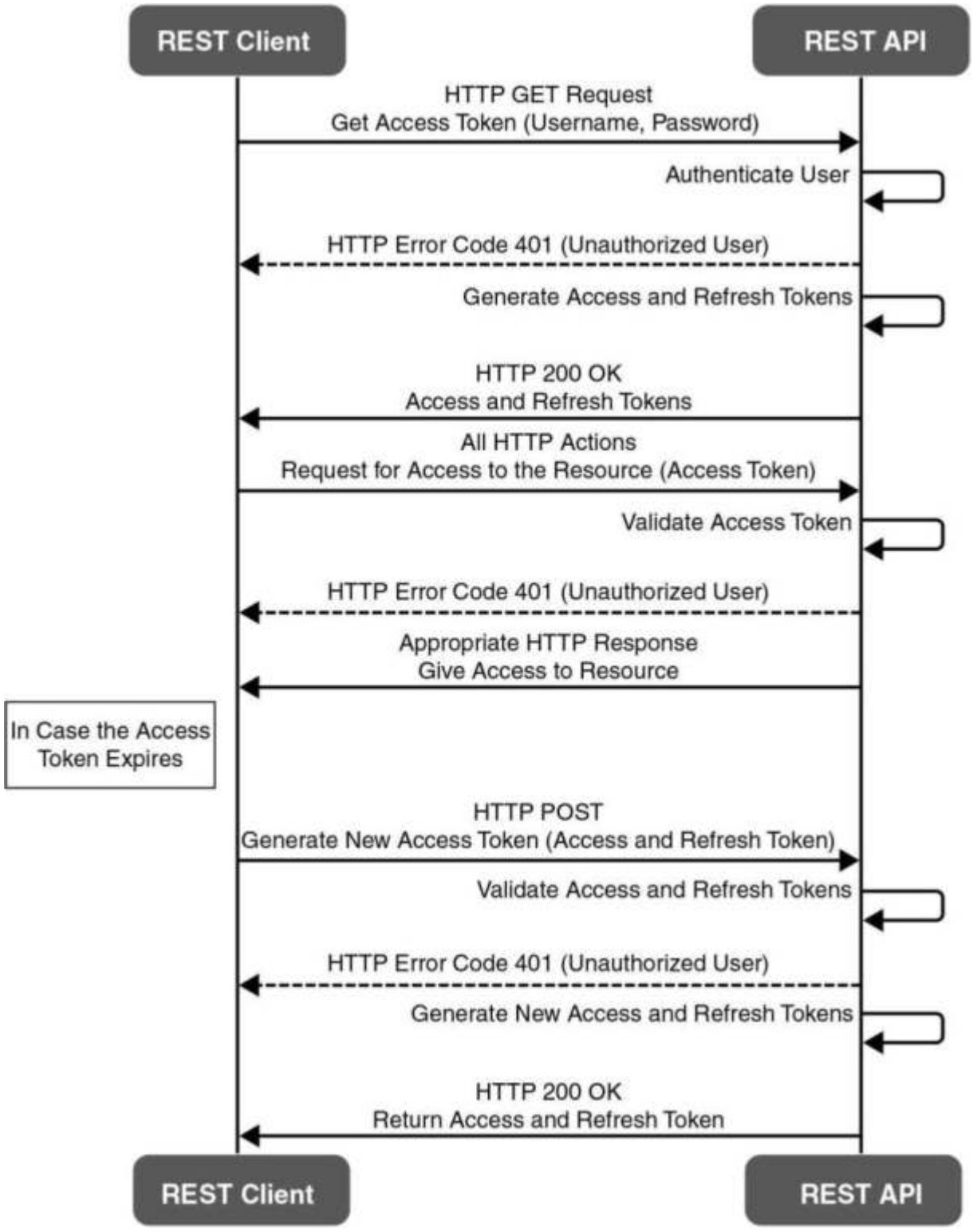


Figure 6-5 *Token-Based Authentication Example*

Simple Object Access Protocol (SOAP)

Simple Object Access Protocol (SOAP) is used to access web services. Although HTTP is the most commonly deployed transport for SOAP, SOAP can use either Simple Mail Transfer Protocol (SMTP) or HTTP. SOAP exchanges data between applications built on different programming languages, such as

Java, .NET, and PHP. SOAP dramatically simplifies a developer's life, eliminating the need to know how to develop in each of these specific programming languages. It makes it possible to exchange data between these applications in a more simplified manner without requiring a developer to be an expert in all the different languages. SOAP is based on XML(Extensible Markup Language). Because most programming languages today have libraries for working with XML, SOAP can act as an intermediary specification between the different applications.

SOAP uses XML to communicate between web services and clients. Because SOAP is platform- and operating system-independent, it can work with both Windows and Linux platforms. SOAP messages, which typically consist of the following four main components, are sent between the web applications and the clients (see Figure 6-6):

- Envelope
- Header
- Body
- Fault (optional)

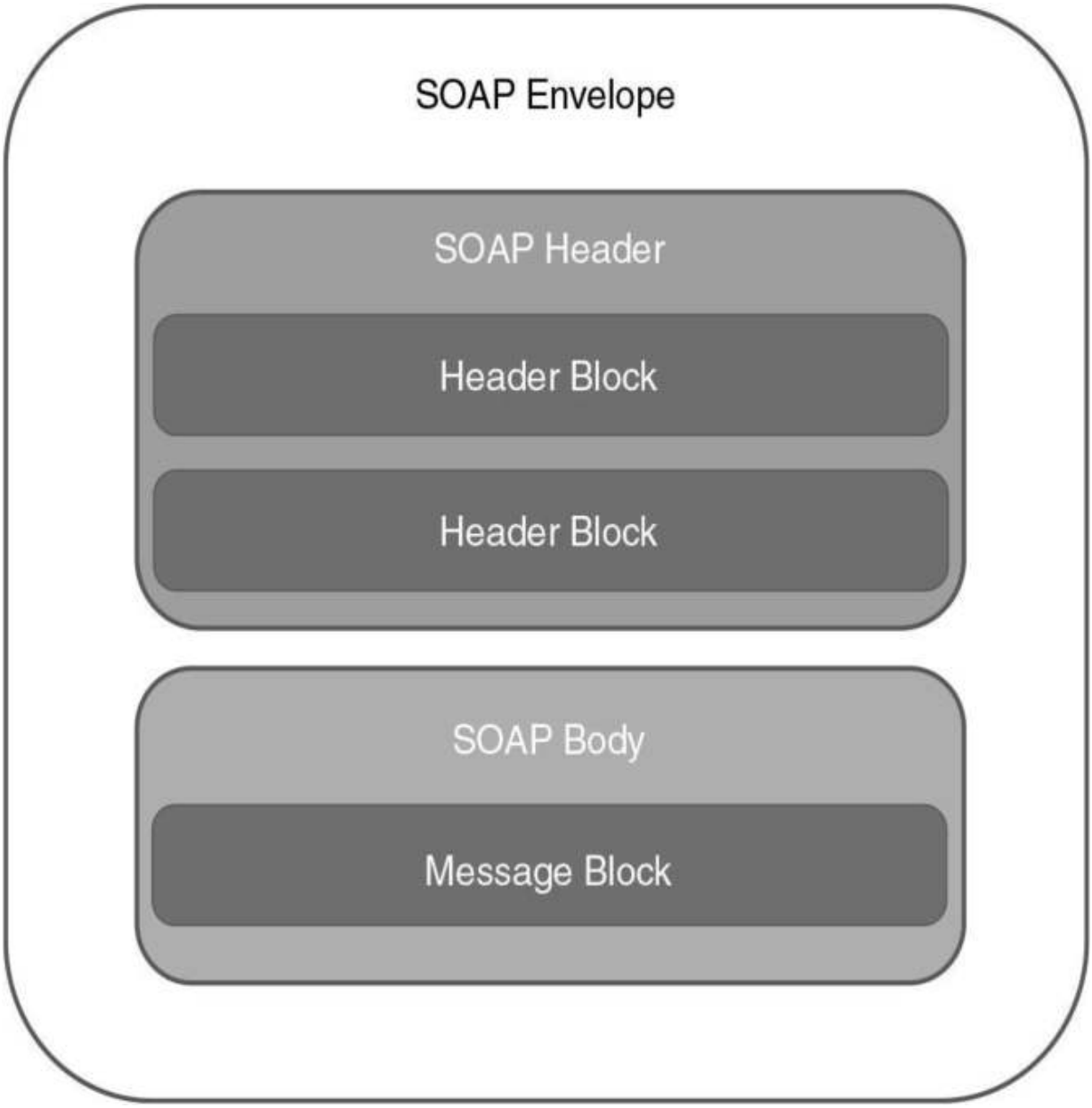


Figure 6-6 *SOAP Message Format*

The SOAP envelope encloses the XML data and identifies it as a SOAP message. The envelope indicates the beginning and the end of the SOAP message. The next portion of a SOAP message is the SOAP header, which can contain multiple header blocks. Header blocks are targeted to specific SOAP receiver nodes. A SOAP message must come before the body element if it contains a header. The SOAP body contains the actual message designated for the SOAP receiver. Every SOAP envelope must contain at least one body element. Typically, SOAP messages are automatically generated by the web service when it's called by the client. Figure 6-7 illustrates the high-level communication that occurs between a client and a server or web service.

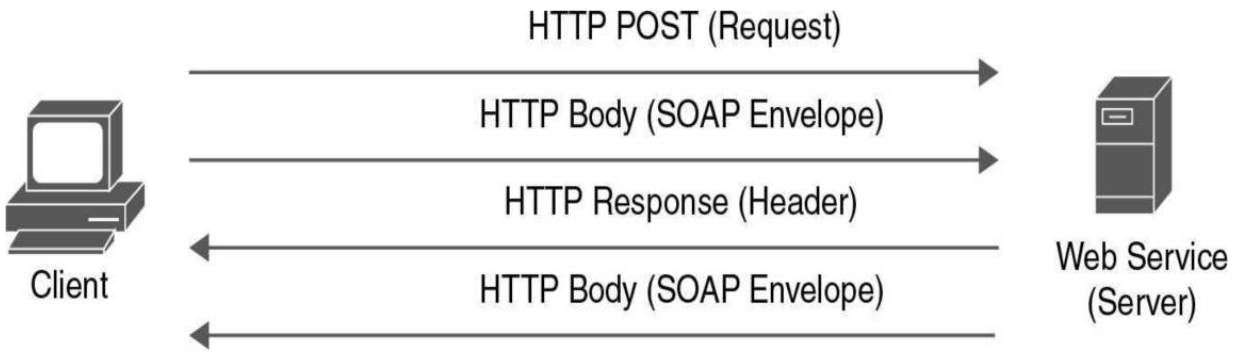


Figure 6-7 *High-Level SOAP Communication*

Another potentially beneficial aspect of SOAP is that, because it primarily uses HTTP, it is efficient in passing through firewalls without requiring additional ports to be allowed or opened for web service traffic. This can save time and reduce some operational overhead. To reiterate, the benefit of SOAP is its capability to work between different languages while using a simple common HTTP and XML structure. Example 6-4 shows a sample SOAP message used to leverage an HTTP GET to retrieve the price for Cisco's stock, using the ticker symbol CSCO.

Example 6-4 *Sample SOAP Message*

```
POST /InStock HTTP/1.1

Host: www.example.org

Content-Type: application/soap+xml; charset=utf-8

Content-Length: 299

SOAPAction: "http://www.w3.org/2003/05/soap-envelope"


<?xml version="1.0"?>

<soap:Envelope xmlns:soap="http://www.w3.org/2003/05/soap-envelope" xmlns:m="http://

www.example.org">

  <soap:Header>

  </soap:Header>

  <soap:Body>

    <m:GetStockPrice>

      <m:StockName>CSCO</m:StockName>

    </m:GetStockPrice>

  </soap:Body>

</soap:Envelope>
```

The World Wide Web Consortium (W3C) recommends using the current SOAP specification, version 1.2. The previous version of SOAP is 1.1. Although it is possible for a SOAP node to support both versions of the protocol, a protocol binding must be used for the version of SOAP that the client intends to use.

As mentioned briefly at the beginning of this section, SOAP can include optional fault messages. Table 6-4 lists the options that are available for the optional fault messages as well as which of them are optional.

Table 6-4 *SOAP Fault Options*

Fault Element	Description	Optional
faultCode	Specifies the fault code of an error	No
faultString	Describes an error	No
faultActor	Specifies who caused a fault	Yes
detail	Applies specific error messages	Yes

Table 6-4 shows the options available in SOAP version 1.2. faultString provides a description of an error message that is generated. This is not an optional field; it is mandatory in the communications between the client and the web service. **faultActor** specifies which node caused a fault. Although this field provides additional information about who caused the fault, it is optional. The detail element provides application-specific error messages; that is, this element is populated with information provided by the application. SOAP fault messages can contain various faultCode options, which indicate what errors were generated and potentially who caused each error (the sender or receiver). Table 6-5 lists the available SOAP fault codes and their associated use cases.

Table 6-5 *SOAP Fault Codes*

SOAP Fault Code	Description
VersionMismatch	The faulting node found an invalid element information item instead of the expected envelope element information item. The namespace, local name, or both did not match the envelope element information item required by this recommendation.
MustUnderstand	This is a child element of the SOAP header. If this attribute is set, any information that was not understood triggers this fault code.
DataEncodingUnknown	A SOAP header block or SOAP body child element information item targeted at the faulting SOAP node is scoped.
Sender	The message was incorrectly formed or did not contain the information needed to succeed. For example, the message might have lacked the proper authentication or payment information. This code generally indicates that the message will not be resent without change.
Receiver	The message could not be processed for reasons attributable to the message's processing rather than the contents of the message itself. For example, processing could include communicating with an upstream SOAP node, which did not respond. The message could succeed, however, if resent at a later point in time.

The fault message shown in Example 6-5 was generated because the Detail value wasn't interpreted correctly due to the typo in the XML <m:MaxTime> P5M</m:MaxTime>. The value P5M caused the issue because the code was expecting 5PM. The XML code and value should be <m:MaxTime>5PM</m:MaxTime> in this case.

Example 6-5 *Sample SOAP Fault*

```
<env:Envelope xmlns:env=”http://www.w3.org/2003/05/soap-envelope”
              xmlns:m="http://www.example.org/timeouts"
              xmlns:xml="http://www.w3.org/XML/1998/namespace">
 <env:Body>
  <env:Fault>
   <env:Code>
     <env:Value>env:Sender</env:Value>
     <env:Subcode>
      <env:Value>m:MessageTimeout</env:Value>
     </env:Subcode>
   </env:Code>
   <env:Reason>
     <env:Text xml:lang="en">Sender Timeout</env:Text>
   </env:Reason>
   <env:Detail>
     <m:MaxTime>P5M</m:MaxTime>
   </env:Detail>
  </env:Fault>
 </env:Body>
</env:Envelope>
```

> **Note**
>
> The examples used in this chapter are all based on SOAP version 1.2.

Remote-Procedure Calls (RPCs)

Remote procedure calls (RPCs) make it possible to execute code or a program on a remote node in a network. RPCs behave as if the code were executed locally on the same node, even though it runs in a remote address space, such as another system on the network. Most remote and local calls are very similar and can be distinguished from one another based on whether they are local or remote. RPCs are sometimes also known as function or subroutine calls. Using an RPC is a very common way to execute specific commands, such as GET or POST operations to a set API or URL.

When a client sends a request message, the RPC translates it and then sends it to the server. A request may be a procedure or a function call destined for a remote server. When a server receives the request, it responds to the client. The client is blocked while this communication is happening, allowing the server time to process the call. Once the call is processed and a response has been sent back to the client, the communication between the client and server is unblocked so the client can resume executing the procedure call. This security mechanism prevents flooding RPCs to brute-force the server and cause denial-of-service (DoS) attacks or resource exhaustion. Figure 6-8 showcases the high-level RPC communications between a client and a server.

As mentioned earlier in this section, an RPC call is blocked during the waiting periods. Once a procedure is executed and the response is sent from the server and received by the client, the execution of the procedure continues. (This means that RPC calls are typically synchronous. There are also asynchronous RPC calls, but this section focuses on synchronous RPC calls.)

Now that the high-level communications of RPC have been covered, let's look at an example of an RPC request message. There are different versions of RPC messages. However, the most common is XML-RPC; XML-RPC was also the most common version before SOAP became available. Example 6-6 shows a simple XML-RPC call that uses a GET to retrieve the name of the 21st state added to the United States.

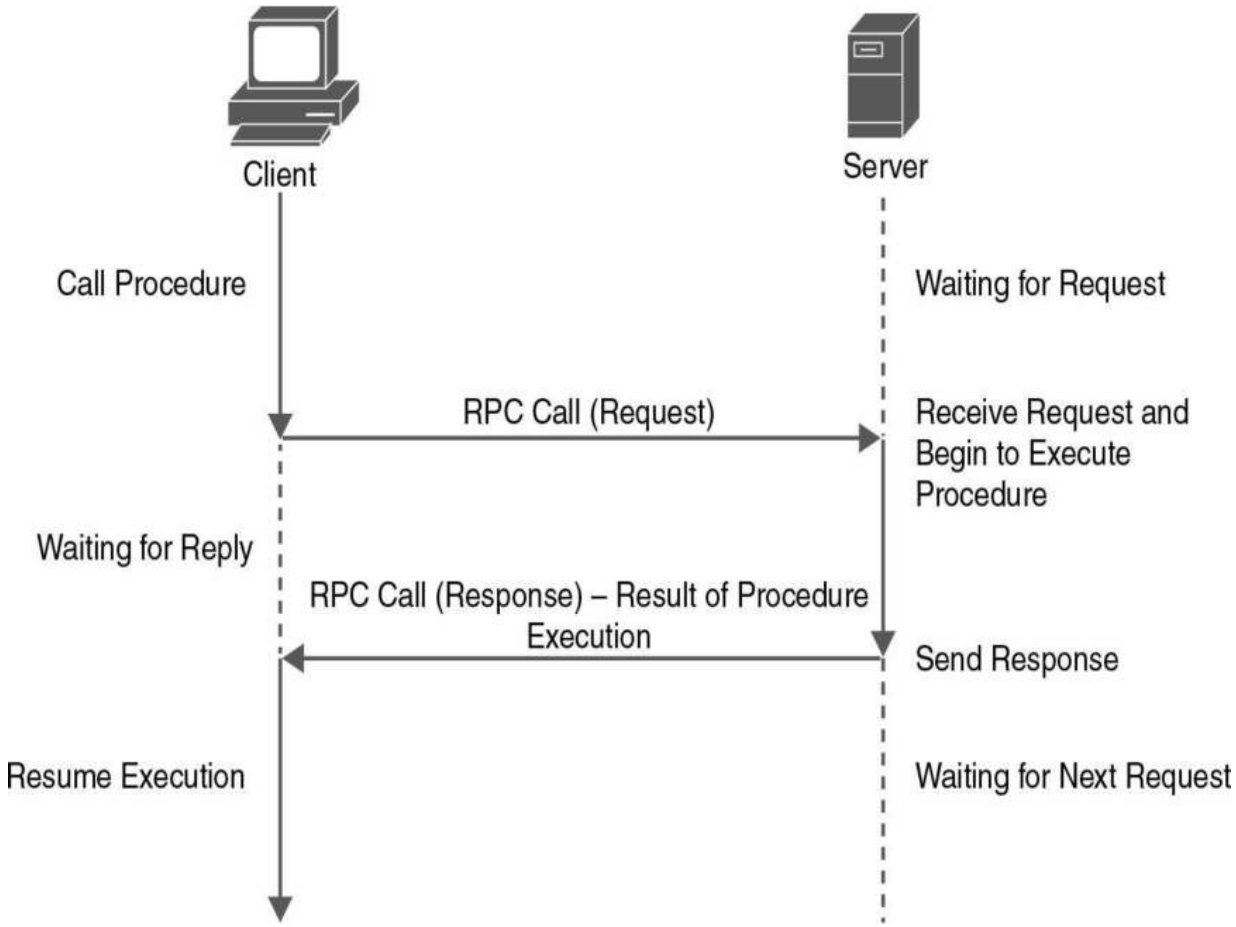


Figure 6-8 *High-Level RPC Communications*

Example 6-6 *Sample XML-RPC Request Message*

```
<?xml version="1.0"?>
<methodCall>
   <methodName>examples.getStateName</methodName>
   <params>
    <param>
         <value><i4>21</i4></value>
     </param>
   </params>
</methodCall>
```

You can see in Example 6-6 that the XML format is very similar to that of SOAP, making these messages simple for humans to read, digest, and build. Example 6-7 shows an XML-RPC reply message, in which the response to the GET message from Example 6-6 is Illinois.

Example 6-7 *Sample XML-RPC Reply Message*

```
<?xml version="1.0"?>
<methodResponse>
  <params>
    <param>
        <value><string>Illinois</string></value>
    </param>
  </params>
</methodResponse>
```

Exam Preparation Tasks

As mentioned in the section "How to Use This Book" in the Introduction, you have a couple of choices for exam preparation: the exercises here, Chapter 19, "Final Preparation," and the exam simulation questions on the companion website.

Review All Key Topics

Review the most important topics in this chapter, noted with the Key Topic icon in the outer margin of the page. Table 6-6 lists these key topics and the page number on which each is found.

Table 6-6 *Key Topics for Chapter 6*

Key Topic Element	Description	Page Number
Section	Synchronous Versus Asynchronous APIs	
Table 6-2	HTTP Functions and Sample Use Cases	
Table 6-3	CRUD Functions and Sample Use Cases	
Section Paragraph	RESTful API Authentication SOAP structure and components	

Define Key Terms

Define the following key terms from this chapter and check your answers in the glossary:

Representational State Transfer (REST) APIs
synchronous APIs
asynchronous APIs
CRUD functions
API keys
Simple Object Access Protocol (SOAP)
remote-procedure calls (RPC)

Chapter 7. RESTful API Requests and Responses

This chapter covers the following topics:

- **RESTful API Fundamentals:** This section covers the basics of RESTful APIs and details operations such as GET, POST, PUT, and DELETE. Other topics include REST headers and data formats such as XML, JSON, and YAML.
- **REST Constraints:** This section covers the six architectural constraints of REST in detail.
- **REST Tools:** This section covers sequence diagrams and tools such as Postman, curl, HTTPie, and the Python Requests library used to make basic REST calls.

Application programming interfaces (APIs) are the foundation of the new generation of software, including networking, cloud, mobile, and Internet of Things (IoT) software. APIs connect pieces of software, "gluing" together any required information components around a system and enabling two pieces of software to communicate.

REST, which stands for Representational State Transfer, refers to a particular style of API building. Most modern services and networking products today rely on REST for their APIs simply because REST is based on HTTP (the protocol that powers nearly all Internet connections). REST is lightweight, flexible, and scalable, and its popularity has grown.

"Do I Know This Already?" Quiz

The "Do I Know This Already?" quiz allows you to assess whether you should read this entire chapter thoroughly or jump to the "Exam Preparation Tasks" section. If you doubt your answers to these questions or your assessment of your knowledge of the topics, read the entire chapter. Table 7-1 lists the major headings in this chapter and their corresponding "Do I Know This Already?" quiz questions. You can find the answers in Appendix A, "Answers to the 'Do I Know This Already?' Quiz Questions."

Table 7-1 *"Do I Know This Already?" Section-to-Question Mapping*

Foundation Topics Section	Questions
RESTful API Fundamentals	1–5
REST Constraints	6, 7
REST Tools	8

Caution

The goal of self-assessment is to gauge your mastery of the topics in this chapter. If you need to know the answer to a question or are only partially sure of the solution, you should mark that question as wrong for self-assessment purposes. Giving yourself credit for an answer that you correctly guess skews your self-assessment results and might provide you with a false sense of security.

1. In HTTP, to make a successful GET request to the server, the client needs to include at least which of the following? (Choose two.)

   a. URL

   b. Method

   c. Headers

   d. Body

2. Which of the following is not an HTTP method?

   a. GET

   b. HEAD

   c. TRIGGER

d. PATCH

3. Webhooks are like which of the following? (Choose two.)

   a. Remote procedure calls

   b. Callback functions

   c. State-altering functions

   d. Event-triggered notifications

4. Which response code indicates that a resource has moved?

   a. 201

   b. 301

   c. 401

   d. 501

5. Which of the following models the interactions between various objects in a single use case?

   a. REST APIs

   b. Sequence diagrams

   c. Excel sheets

   d. Venn diagrams

6. Which REST API architectural constraint allows you to download and execute code?

   a. Client/server

   b. Statelessness

   c. Code on demand

   d. Layered systems

7. Rate limiting is an essential REST API design method for developers. Rate-limiting techniques are used to _____.

   a. increase security

   b. have business impact

   c. enhance efficiency end-to-end

   d. do all the above

8. To add HTTP headers to a Python request, you can simply pass them in as which of the following?

   a. list

   b. dict

   c. tuple

   d. set

Foundation Topics

RESTful API Fundamentals

An **application programming interface** (API) is a set of functions and procedures intended to be used as an interface for software components to communicate with each other. An API may be for a web app, an operating system, a database system, computer hardware, or a software library. A typical example of an API is the Google Maps API, which lets you interface with Google Maps so that you can display maps in your application, query locations, and so on. Figure 7-1 shows a simple way to visualize an API.

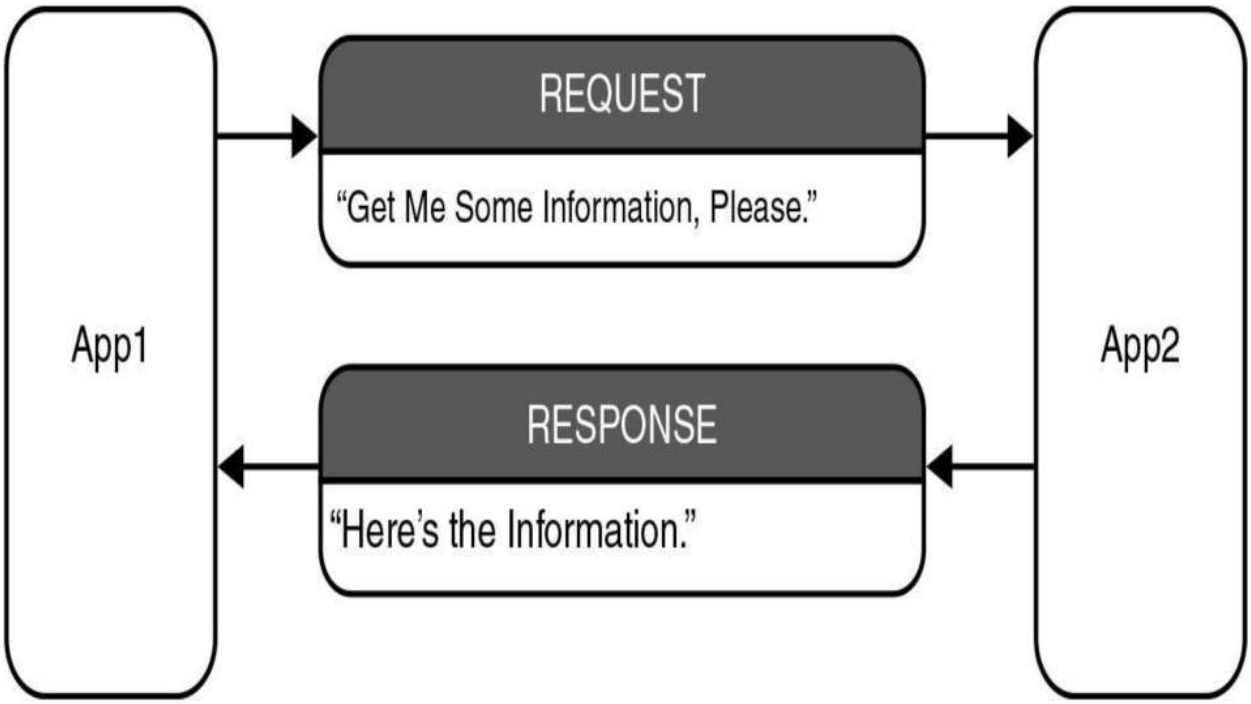


Figure 7-1 *APIs are a contract between two communicating applications*

The following sections look at the different API types.

API Types

APIs can be broadly classified into three categories based on the type of work that each one provides:

- **Service API:** In a service API, an application can call on another to solve a particular problem (see Figure 7-2). Usually, these systems can exist independently. For example, in a payment system, an application can call the API to accept payments via credit cards. As another example, an application can call an API to validate and authenticate users with a user-management system.

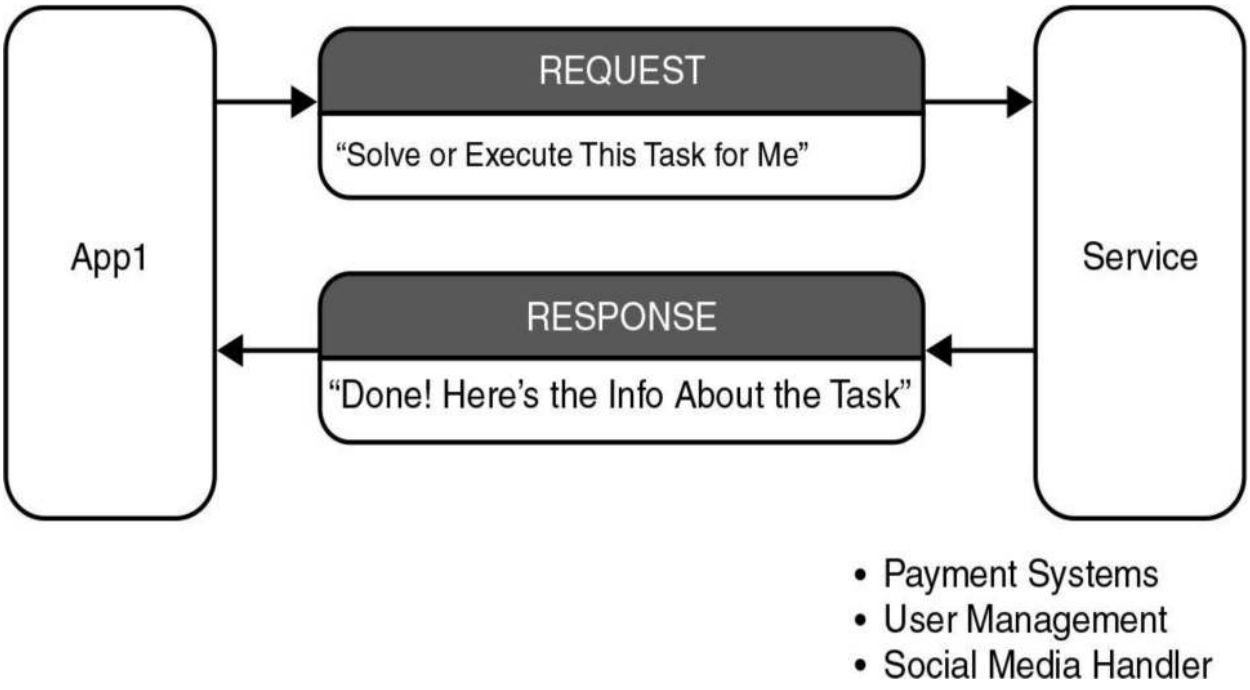


Figure 7-2 *Service API Providing a Complete Service to the Calling Application*

- **Information API:** An information API allows one application to ask another for information. *Information* in this context can refer to data gathered over time, telemetry data, or a list of currently connected devices. Figure 7-3 provides a visual representation of an information API.

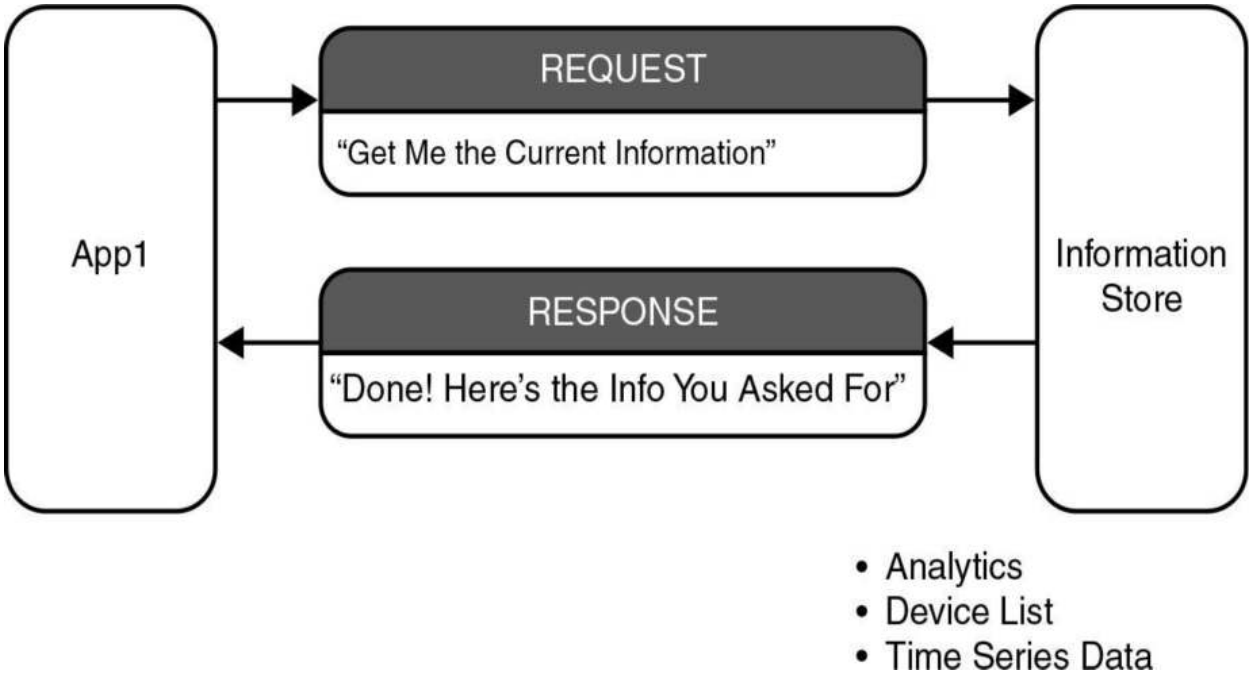


Figure 7-3 *Information API Providing Information or Analysis of Data That Has Been Collected*

- **Hardware API:** Application developers use hardware APIs to gain access to the features of hardware devices. Usually, these APIs encompass some kind of hardware or sensors, and an application can call this kind of API to get the GPS location or real-time sensor data such as temperature or humidity. Figure 7-4 visually represents what a hardware API does.

API Access Types

There are typically three ways APIs can be accessed:

- **Private:** A private API is for internal use only. This access type gives a company the most control over its API.
- **Partner:** A partner API is shared with specific business partners. This can provide additional revenue streams without compromising quality.
- **Public:** A public API is available to everyone. This allows third parties to develop applications that interact with an API and can be a source for innovation.

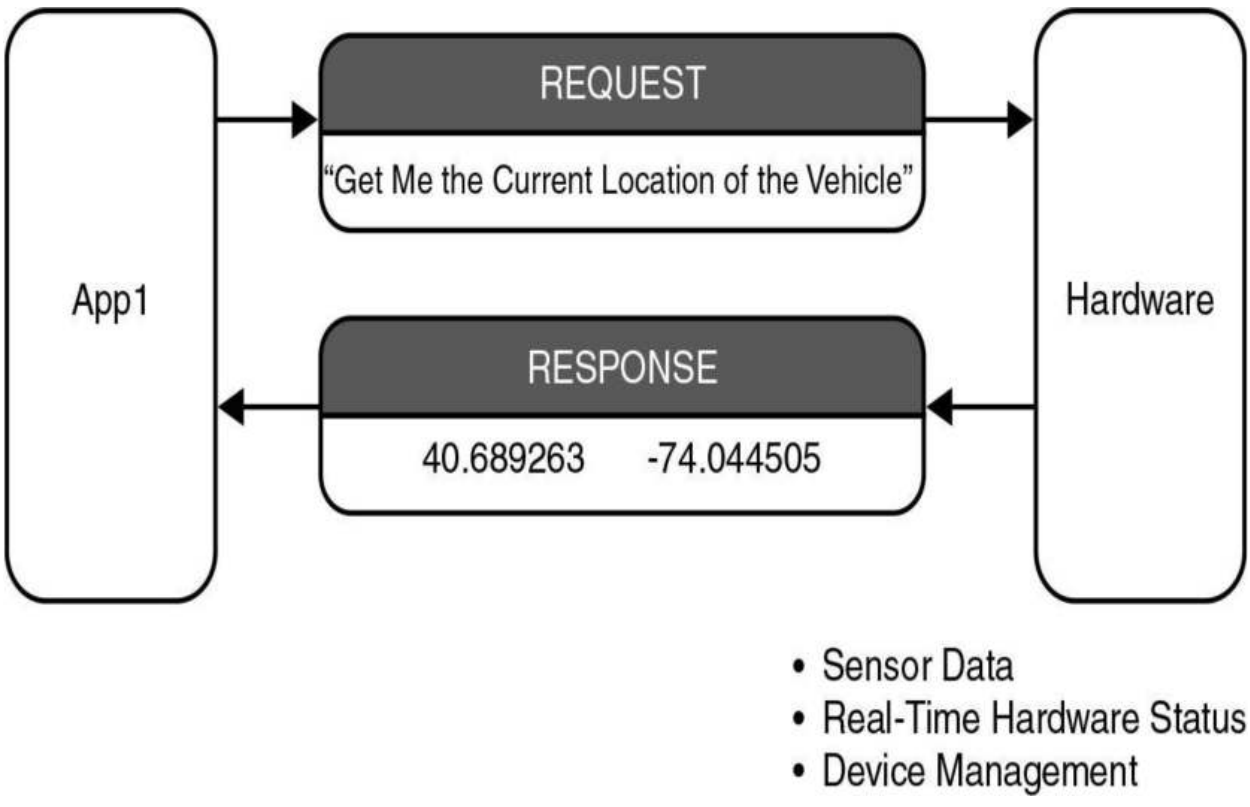


Figure 7-4 *Hardware API Providing Access to Hardware to Get or Set Data*

APIs are designed to interact through a network regardless of how they are accessed. Because the Internet is the most widely used communications network, most APIs are designed based on web standards. Not all remote APIs are web APIs, but it's fair to assume they are remote.

Thanks to the ubiquity of HTTP on the web, most developers have adopted it as the protocol underlying their APIs. The most significant benefit of using HTTP is that it reduces the learning curve for developers, which encourages the use of an API. HTTP has several valuable features in building a good API, which will be apparent as we explore the basics in the next section.

HTTP Basics

A web browser is a classic example of an HTTP client. Communication in HTTP centers around a concept called the request/response cycle, in which the client sends the server a request to do something. The server, in turn, sends the client a response saying whether or not the server can do what the client asked. Figure 7-5 provides a very simple illustration of how a client requests data from a server and how the server responds to the client.

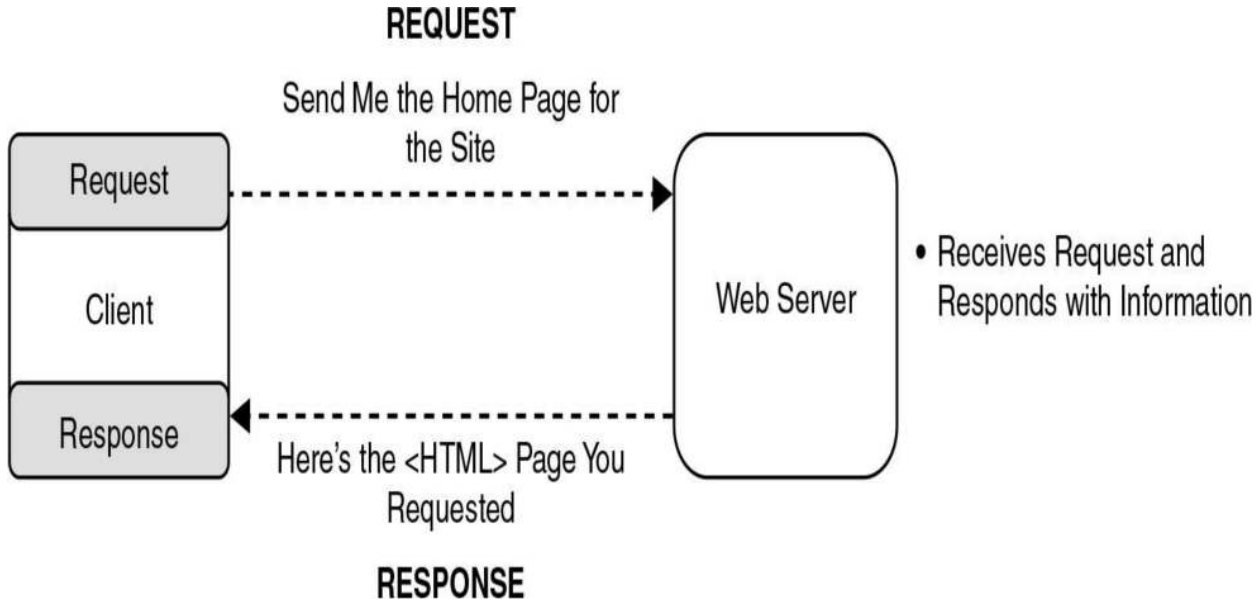


Figure 7-5 *Simple HTTP Request/Response Cycle*

Now let's look at a request from the HTTP point of view, where the client (web browser) makes a request (GET /index.hml) to the server (developer.cisco.com). The server eventually responds to the client with the actual HTML page, which then gets rendered by the browser. Figure 7-6 provides a very simple representation of how a client sends a GET request requesting the page from the server and how the server responds with the HTML page to the client.

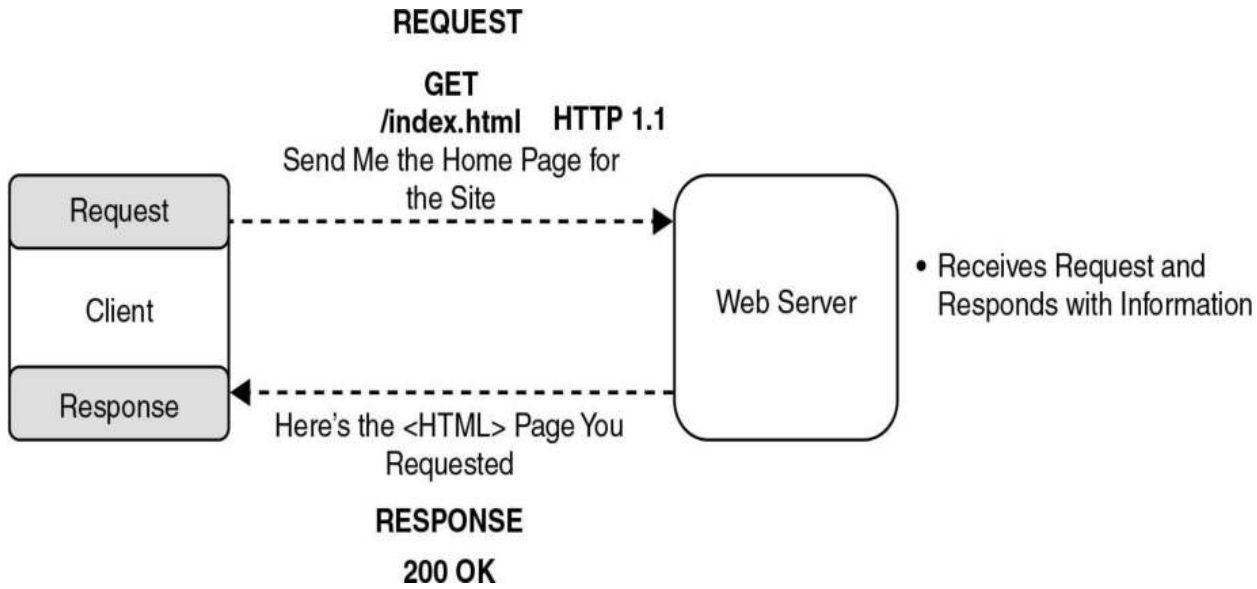


Figure 7-6 *Simple HTTP GET Request with 200 OK Response*

In HTTP, to make a successful request to the server, the client needs to include four items:

- URL (uniform resource locator)
- Method
- List of headers
- Body

The following sections look at each of these items in detail.

Uniform Resource Locator (URL)

A URL is similar to a house address in that it defines the location where a service resides on the Internet. A URL typically has four components, as shown in Figure 7-7:

- Protocol
- Server/host address
- Resource
- Parameters

These four components are shown in Figure 7-7.

As you can see, the server or host address is the unique server name, /api/rooms/livingroom defines a resource to access, and lights?state=ON is the parameter to send to take some action.

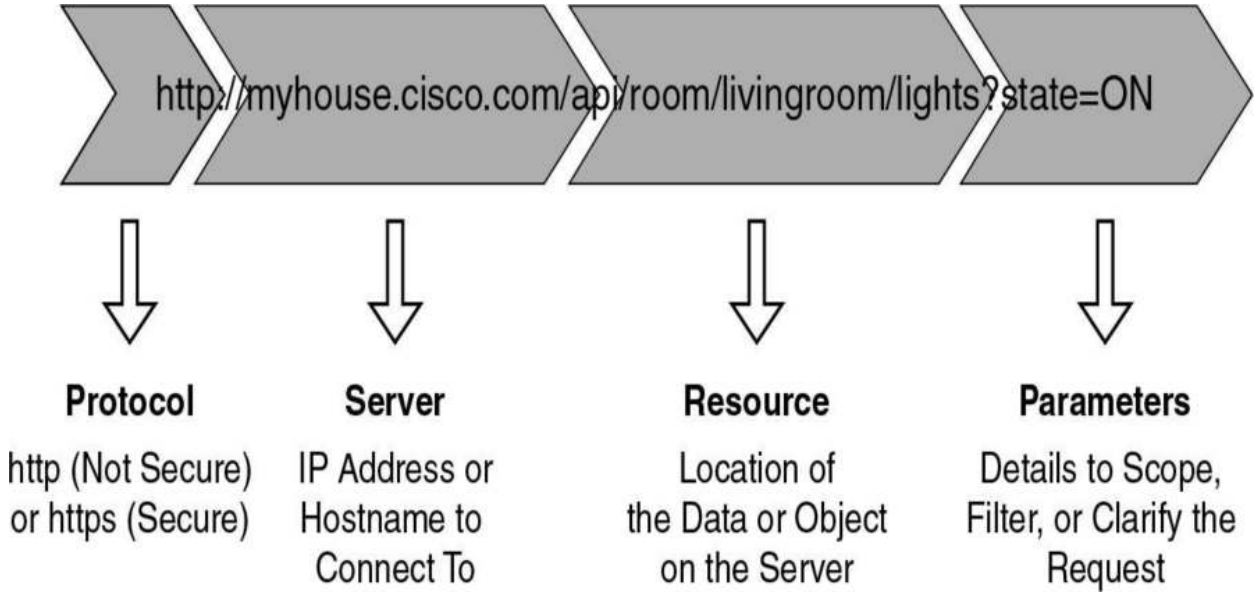


Figure 7-7 *Anatomy of an HTTP URL*

Method

HTTP defines a set of request methods, outlined in Table 7-2. A client can use one of these request methods to send a request message to an HTTP server.

Table 7-2 *Request Methods*

Method	Explanation
GET	A client can use a GET request to get a web resource from the server.
HEAD	A client can use a HEAD request to get the header that a GET request would have obtained. Because the header contains the last modified date of the data, it can be used to check against the local cache copy.
POST	A client can use a POST request to post data or add new data to the server.
PUT	A client can use a PUT request to ask a server to store or update data.
PATCH	A client can use a PATCH request to ask a server to partially store or update data.
DELETE	A client can use a DELETE request to request a server delete data.
TRACE	A client can use a TRACE request to request a server return a diagnostic trace of its actions.
OPTIONS	A client can use an OPTIONS request to ask a server to return a list of the request methods it supports.
CONNECT	A client can use a CONNECT request to tell a proxy to connect to another host and simply reply with the content without attempting to parse or cache it. This request is often used to make a TLS connection through the proxy.

REST Methods and CRUD

As you have seen, REST is an architectural paradigm that allows developers to build RESTful services. These RESTful applications use HTTP requests to handle all four CRUD operations: CREATE, READ, UPDATE, and DELETE. These four operations are the operations most commonly used in manipulating data. The HTTP methods map one-to-one to the CRUD operations, as shown in Table 7-3.

Table 7-3 *Mapping HTTP Methods to CRUD Operations*

HTTP Method	Operation	Explanation
POST	CREATE	Used to create a new object or resource. Example: Add a new room to a house
GET	READ	Used to retrieve resource details from the system. Example: Get a list of all the rooms or all the details of one room
PUT	UPDATE	Typically used to replace or update a resource. Can be used to modify or create a resource. Example: Update details of a room
PATCH	UPDATE	Used to modify some details about a resource. Example: Change the dimensions of a room
DELETE	DELETE	Used to remove a resource from the system. Example: Delete a room from a house.

Deep Dive into GET and POST

GET is the most common HTTP request method. A client can use the GET request method to request (or "get") a resource from an HTTP server. GET requests, which have unique qualities, fetch information, and that's it; they have no side effects, make no modifications to the system, create nothing, and destroy nothing. GET requests should, in other words, be safe and idempotent. (Idempotent means that no matter how often you act, the state of the system you're dealing with remains the same.)

A GET request message has the following components, as shown in Figure 7-8:

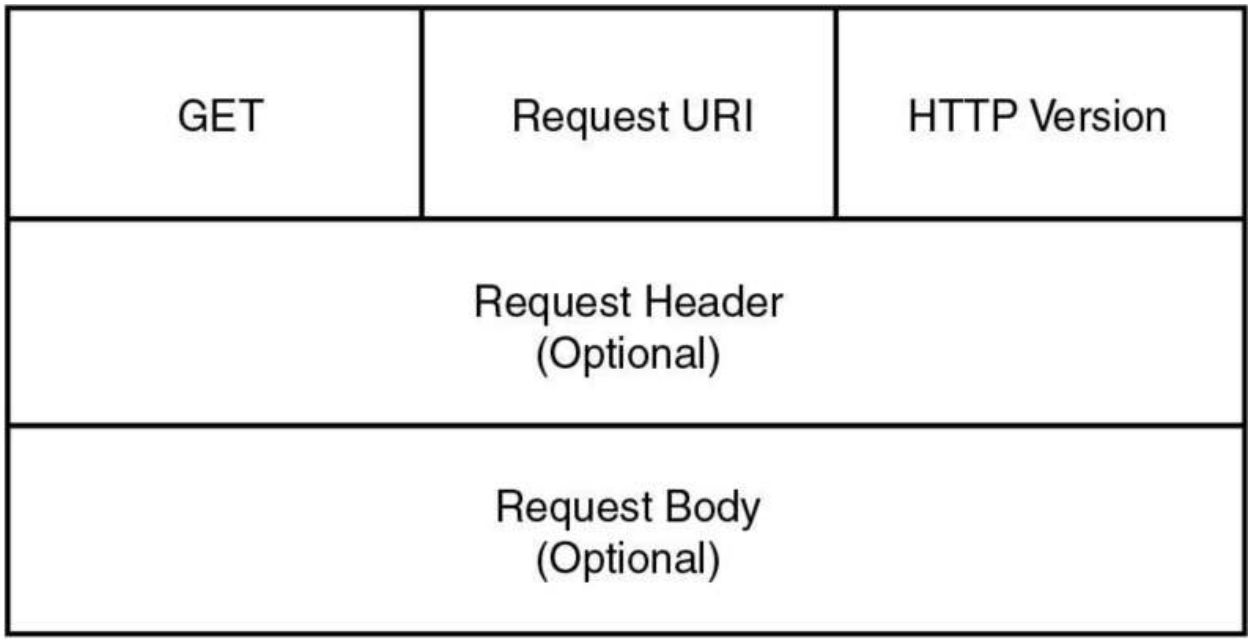


Figure 7-8 *Syntax of a GET Request*

- **GET:** The keyword GET must be all uppercase.
- **Request URI:** Specifies the path of the resource requested, which must begin from the root / of the document base directory.
- **HTTP version:** Either HTTP/1.0 or HTTP/1.1. This client negotiates the protocol to be used for the current session. For example, the client may request to use HTTP/1.1. If the server does not support HTTP/1.1, it may inform the client in response to using HTTP/1.0.
- **Request headers (optional):** The client can use optional request headers (such as accept and accept-language) to negotiate with the server and ask the server to deliver the preferred contents (such as in the language the client prefers).
- **Request body (optional):** A GET request message has an optional request body containing the query string (explained later in this chapter).

The POST request method posts additional data to the server (for example, submitting HTML form data or uploading a file). Issuing an HTTP URL from the browser always triggers a GET request. To activate a POST request, use an HTML form with attribute method="post" or write your code. The POST request is the same as the GET request for submitting HTML form data except that the URL-encoded query string is sent in the request body rather than appended behind the request URI.

The POST request has the following components, as shown in Figure 7-9:

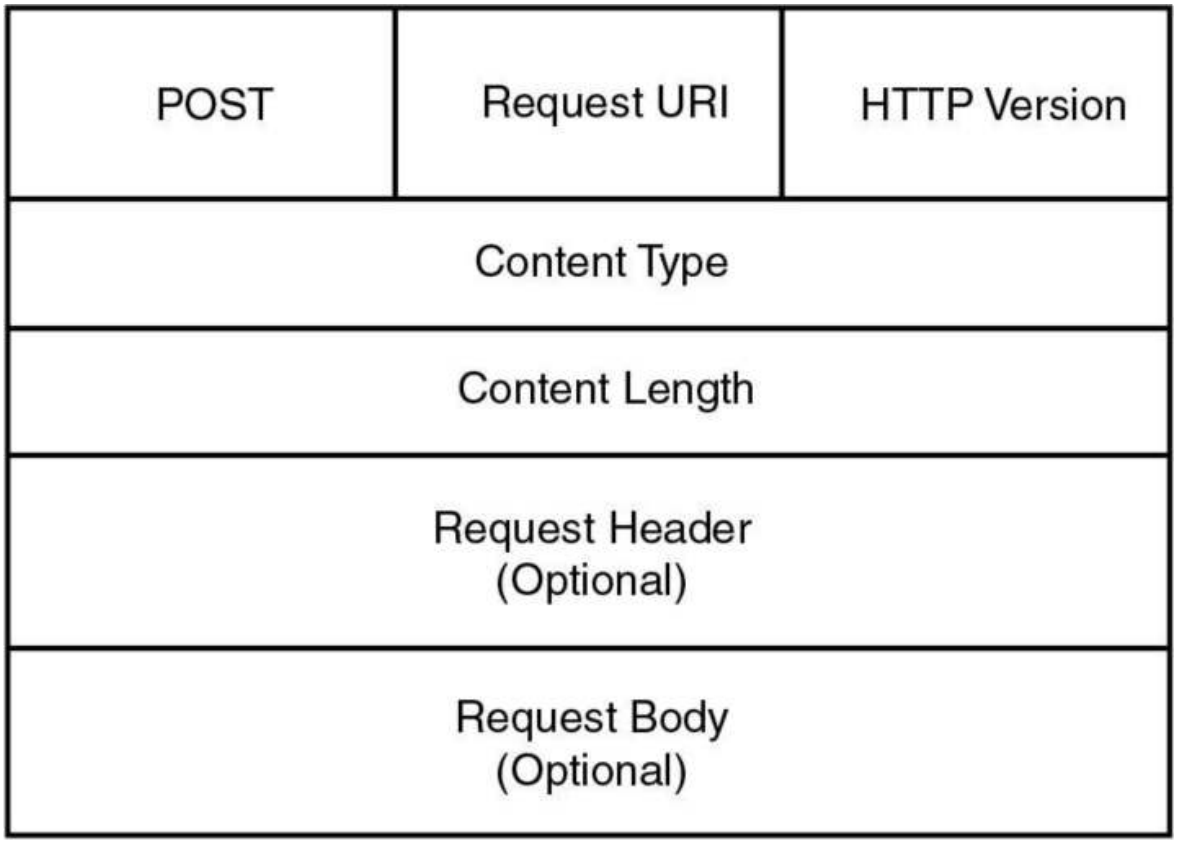


Figure 7-9 *Syntax of a POST Request*

- **POST:** The keyword POST must be all uppercase.
- **Request URI:** Specifies the path of the resource requested, which must begin from the root / of the document base directory.
- **HTTP version:** Either HTTP/1.0 or HTTP/1.1. This client negotiates the protocol to be used for the current session. For example, the client may request to use HTTP/1.1. If the server does not support HTTP/1.1, it may inform the client in response to using HTTP/1.0.
- **Request headers (optional):** The client can use optional request headers, such as content type and content length, to inform the server of the media type and the size of the request body, respectively.
- **Request body (optional):** A POST request message has an optional request body containing the query string (explained later in this chapter).

HTTP Headers

The HTTP headers and parameters provide a lot of information that can help you trace issues when you encounter them. HTTP headers are an essential part of an API request and response as they represent the metadata associated with the API request and response. Headers carry information for the following:

- Request and response body
- Request authorization
- Response caching
- Response cookies

In addition, HTTP headers have information about HTTP connection types, proxies, and so on. Most headers manage connections between a client, a server, and proxies.

Headers are classified as request headers and response headers. When sending a request API, you have to set the request headers and the assertion against the response headers to ensure that the correct headers are returned.

Request Headers

The request headers appear as name: value pairs. Multiple values, separated by commas, can be specified as follows:

```
request-header-name: request-header-value1, request-header-value2, ...
```

The following are some examples of request headers:

```
Host: myhouse.cisco.com
Connection: Keep-Alive
Accept: image/gif, image/jpeg, */*
Accept-Language: us-en, fr, cn
```

Response Headers

The response headers also appear as name: value pairs. As with request headers, multiple values can be specified as follows:

```
response-header-name: response-header-value1, response-header-value2, ...
```

The following are some examples of response headers:

```
Content-Type: text/html
Content-Length: 35
Connection: Keep-Alive
Keep-Alive: timeout=15, max=100
The response message body contains the resource data requested.
```

The following are some examples of request and response headers:

- **Authorization:** Carries credentials containing the client's authentication information for the resource being requested.
- **WWW-Authenticate:** This is sent by the server if it needs a form of authentication before responding to the resource being requested. It is often sent with response code 401, which means "unauthorized."
- **Accept-Charset:** This request header tells the server which character sets are acceptable to the client.
- **Content-Type:** This header indicates the media type (text/HTML or application/JSON) of the client request sent to the server by the client, which helps process the request body correctly.
- **Cache-Control:** This header is the cache policy defined by the server. For this response, the client can store a cached response and reuse it until the time is defined in the Cache-Control header.

Response Codes

The first line of a response message (that is, the status line) contains the response status code, which the server generates to indicate the outcome of the request. Each status code is a three-digit number:

- **1xx (informational):** The request was successfully received; the server is continuing the process.
- **2xx (success):** The request was received, understood, accepted, and serviced.
- **3xx (redirection):** To complete the request, further action must be taken.
- **4xx (client error):** The request cannot be understood or is unauthorized, or the requested resource could not be found.
- **5xx (server error):** The server failed to fulfill a request.

Table 7-4 describes some commonly encountered status codes.

Table 7-4 *HTTP Status Codes*

Status Code	Meaning	Explanation
100	Continue	The server received the request and is in the process of giving the response.
200	Okay	The request is fulfilled.
301	Move permanently	The resource requested has been permanently moved to a new location. The URL of the new location is given in the Location response header. The client should issue a new request to the new site, and the application should update all references to this new location.
302	Found and redirected (or moved temporarily)	This is the same as code 301, but the new location is temporary. The client should issue a new request, but applications need not update the references.
304	Not modified	In response to the if-modified-since conditional GET request, the server notifies that the resource requested has not been modified.
400	Bad request	The server could not interpret or understand the request; the request message probably has a syntax error.
401	Authentication required	The requested resource is protected and requires the client's credentials (username and password). The client should resubmit the request with the appropriate credentials (username and password).
403	Forbidden	The server refuses to supply the resource, regardless of the client's identity.

404	Not Found	The requested resource cannot be found on the server.
405	Method not allowed	The request method used (for example, POST, PUT, DELETE) is a valid method. However, the server does not allow that method for the resource requested.
408	Request timeout	The request sent to the server took longer than the website's server was prepared to wait.
414	Request URI too large	The URI requested by the client is longer than the server is willing to interpret.
500	Internal server error	The server is confused; this may be caused by an error in the server-side program responding to the request.
501	Method not implemented	The request method is invalid; this could be caused by a typing error, such as Get in place of GET.
502	Bad gateway	The proxy or gateway indicates that it received a bad response from the upstream server.
503	Service Unavailable	The server cannot respond due to overloading or maintenance. The client can try again later.
504	Gateway timeout	The proxy or gateway indicates that it received a timeout from an upstream server.

Now that we have looked at HTTP methods and return codes, let's look at the data sent or received during a GET or POST. Let's look at an example and see how the same information is represented in the various data types. For this example, refer to Figure 7-7. In this example, you are making a GET request to the server at myhouse.com to change the state of the lights in the living room to ON.

The data sent and received in a RESTful connection requires structured data formatting. For the house example, you now see a response from the server that includes information about the house. Standard data formats include XML, JSON, and YAML, described in the following sections.

XML

Extensible Markup Language (XML) is a markup language that encodes information between descriptive tags. XML is a superset of Hypertext Markup Language (HTML), originally designed to describe the formatting of web pages served by servers through HTTP. The encoded information is defined within user-defined schemas that enable any data to be transmitted between systems. An entire XML document is stored as text, both machine-readable and human-readable.

Example 7-1 shows a sample XML response document. As you can see, with XML, you can assign some meaning to the tags in the document. You can extract the various attributes from the response by simply locating the content surrounded by <study_room> and </study_room>; this content is technically known as the <study_room> *element*.

Example 7-1 *XML Data Format*

```
<?xml version="1.0" encoding="UTF-8" ?>
<root>
  <home>this is my house</home>
  <home>located in San Jose, CA</home>
  <rooms>
    <living_room>true</living_room>
    <kitchen>false</kitchen>
    <study_room>
      <size>20x30</size>
    </study_room>
    <study_room>
      <desk>true</desk>
    </study_room>
    <study_room>
      <lights>On</lights>
    </study_room>
  </rooms>
</root>
```

JSON

JSON, short for **JavaScript Object Notation**, is pronounced like the name "Jason." The JSON format is derived from JavaScript object syntax but is entirely text-based. It is a key value data format typically rendered in curly braces {} and square brackets []. JSON is readable, lightweight, and easy for humans to understand.

A key/value pair has a colon (:) that separates the key from the value, and each such pair is separated by a comma in the document or the response.

JSON **keys** are valid strings. The value of a key is one of the following data types:

- String
- Number
- Object
- Array
- Boolean (true or false)
- Null

JSON data consists of:

- Objects: Defined within curly braces {}. Objects contain multiple name/value pairs.
- Arrays: Enclosed in square brackets []. Arrays can contain objects or other serializable values.
- Name/Value Pairs: Each pair consists of a field name (in double quotes) followed by a colon and a value.
- Valid Data Types: JSON values can be strings, numbers, objects, arrays, booleans (true or false), or null.

Example 7-2 shows a sample JSON response document; you can see the complete response. Suppose you are interested in seeing the status of the lights in the study room. In that case, you look at all the values that are present and follow the various key/value pairs (such as "lights": "On") and extract the multiple values from the response by locating the correct keys and corresponding values.

Example 7-2 *JSON Data Format*

```
{

  "home": [
    "this is my house",
    "located in San Jose, CA"
  ],
  "rooms": {
    "living_room": "true",
    "kitchen": "false",
    "study_room": [
      {
        "size": "20x30"
      },
      {
        "desk": true
      },
      {
        "lights": "On"
      }
    ]
}
```

YAML

YAML is an acronym for "YAML Ain't Markup Language." According to the official YAML site (https://yaml.org), "YAML is a human-friendly data serialization standard for all programming languages."

YAML is a data serialization language designed for human interaction. It's a strict superset of JSON, another data serialization language. But because it's a strict superset, it can do everything JSON can and more. One significant difference is that newlines and indentations mean something in YAML, whereas JSON uses brackets and braces to convey similar ideas. YAML uses three main data formats:

- **Scalars:** The simplest is a *key-value* view.
- **Lists/sequences:** Data can be ordered by indexes.
- **Dictionary mappings:** These are similar to scalars but can contain nested data, including other data types.

> **Note:**
> YAML uses indentation (spaces or tabs) to define nesting levels. Newlines matter in YAML; they indicate the start of a new item or key-value pair.

Use Cases for YAML:

- Configuration files for applications, services, and tools.
- Data exchange between different systems.
- Human-readable data representation.

Example 7-3 shows a sample YAML response document. In this example:

- home contains an array of strings.
- rooms is a dictionary with keys like living_room, kitchen, and study_room.
- The study_room section includes nested data (size, desk, and lights)

As you can see, the response is very straightforward and human-readable. If you are interested in seeing the status of the lights in study_room, you can find the study_room section and then look for the value of lights.

Example 7-3 *YAML Data Format*

```
---
home:
- this is my house
- located in San Jose, CA
rooms:
   living_room: 'true'
   kitchen: 'false'
   study_room:
   - size: 20x30
   - desk: true
   - lights: 'On'
```

Webhooks

Webhooks are user-defined HTTP callbacks. A webhook is triggered by an event, such as pushing code to a repository or typing a keyword in a chat window. An application implementing webhooks sends a POST message to a URL when a specific event happens. Webhooks are also referred to as *reverse APIs*, but more accurately, a webhook lets you skip the request step in the request/response cycle. No request is required for a webhook, and a webhook sends data when triggered.

For security reasons, the REST service may perform some validation to determine whether the receiver is valid. A simple validation handshake performs validation, but this is just one way of validating.

The validation token is a unique token specified by the server. Validation tokens can be generated or revoked on the server side through the configuration UI. When the server sends data to a webhook URL, it includes a validation token in the request HTTP header. The webhook URL should have the same validation token value in the HTTP response header. In this way, the server knows it is sending to a validated endpoint, not a rogue one. Figure 7-10 illustrates the flow of webhooks.

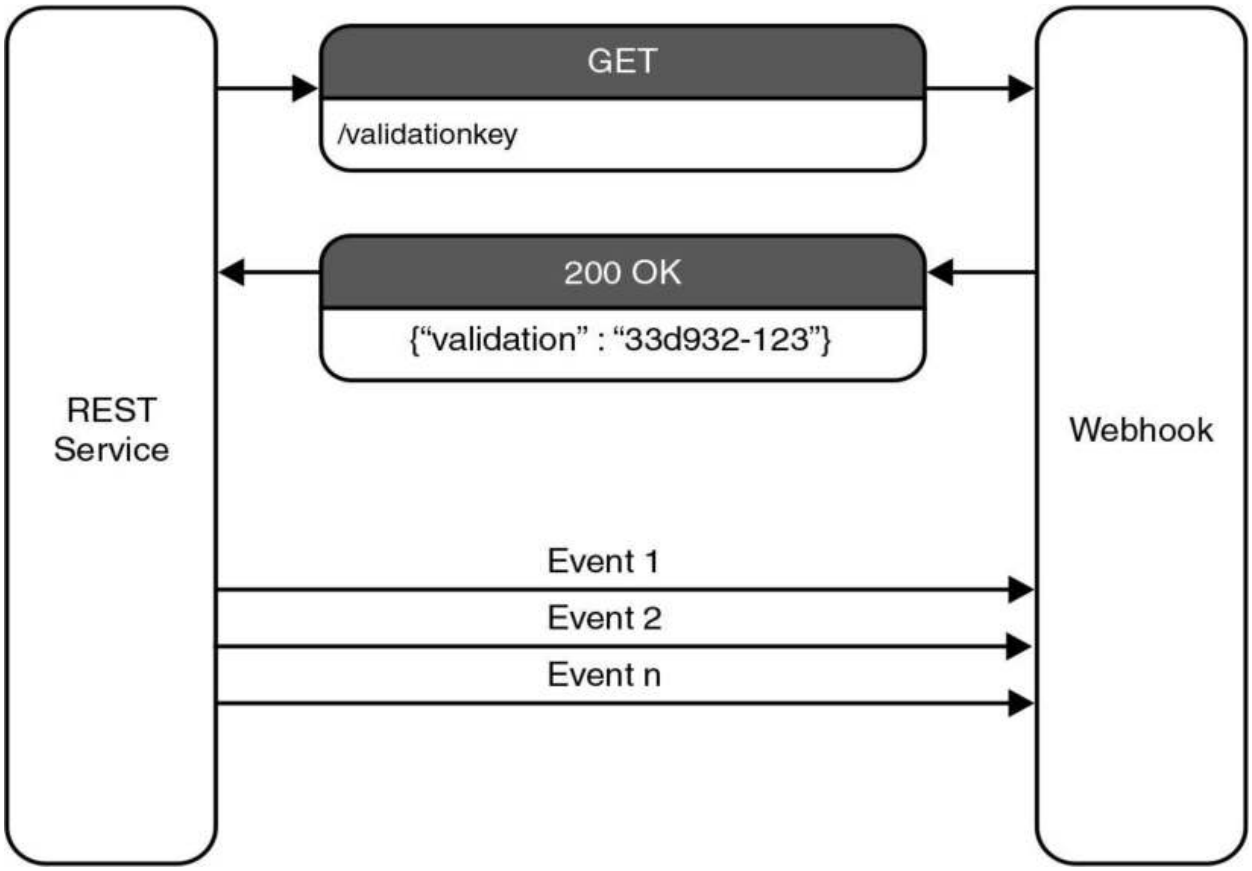


Figure 7-10 *Webhook Validation and Event Flow*

Tools Used When Developing with Webhooks

You will face a particular difficulty when developing an application that consumes webhooks. When using a public service that provides webhooks, you need a publicly accessible URL to configure the webhook service. Typically, you develop on localhost, and the rest of the world cannot access your application, so how do you test your webhooks? ngrok (http://ngrok.com) is a free tool that allows you to tunnel from a public URL to your local application.

Sequence Diagrams

Now that you understand the fundamentals of REST API (request, response, and webhooks), authentication, data exchange, and constraints with rest APIs, it's time to introduce sequence diagrams. A sequence diagram models the interactions between various objects in a single use case. It illustrates how the different parts of a system interact with each other to carry out a function and the order in which the interactions occur when a particular use case is executed. In simpler terms, a sequence diagram shows how different parts of a system work in a sequence to get something done.

Figure 7-11 is a sequence diagram for the example we've been looking at, where a user wants to get a list of all house rooms. For this example, assume that there is a web application with a user interface that renders the list of all the rooms and the various attributes of the rooms.

The sequence of events that occur is as follows:

1. The client browser points to http://myhouse.cisco.com/ (the HTTP GET request sent), which is the web application.
2. The server sends out a REST API request to get all the rooms to the back-end service (/API/getallrooms) to get all the details of the house.
3. The back-end API service returns data in JSON format.
4. The web application processes the JSON and renders the data in the user interface.
5. The client sees the data.

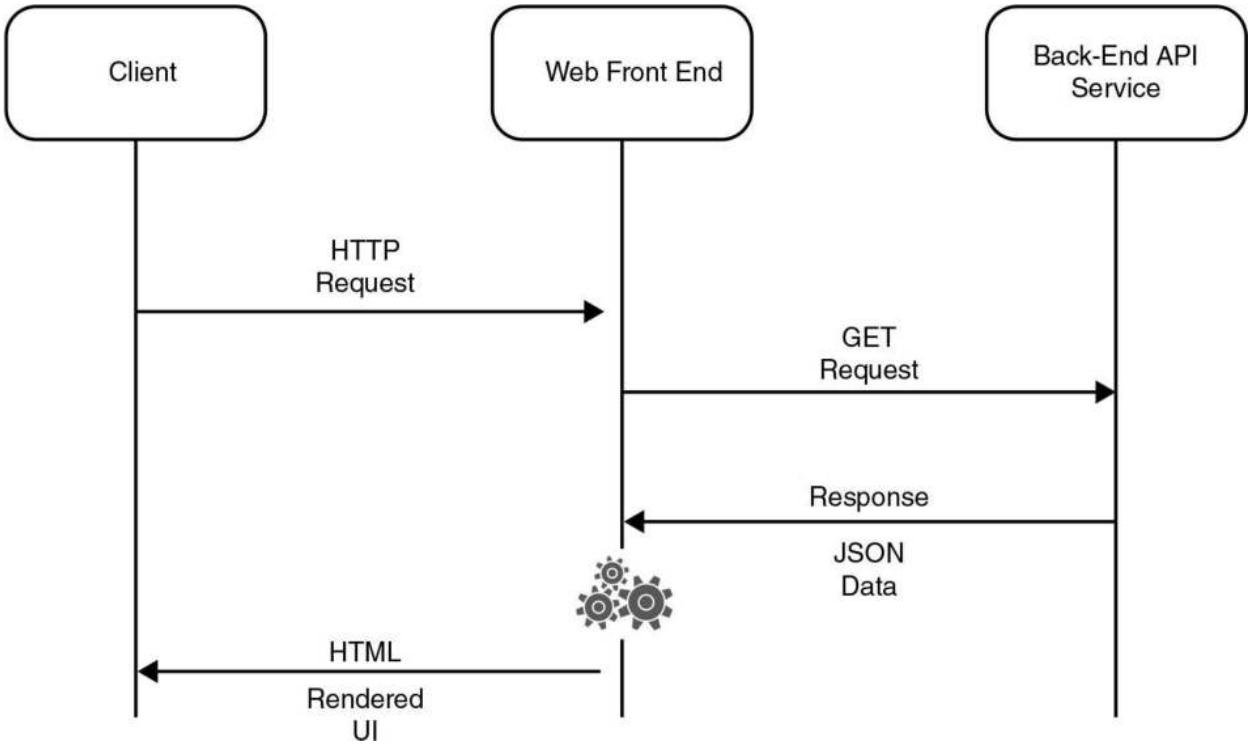


Figure 7-11 *Sequence Diagram Showing End-to-End Flow*

REST Constraints

REST defines six architectural constraints that make any web service a truly RESTful API. These are constraints, also known as Fielding's constraints (see https://www.ics.uci.edu/~fielding/pubs/dissertation/top.htm). They generalize the web's architectural principles and represent them as a framework of rules or an architectural style. These are the REST constraints:

- Client/server
- Stateless
- Cache
- Uniform interface
- Layered system
- Code on demand

The following sections discuss these constraints in some detail.

Client/Server

The client and server exist independently. They must have no dependency of any sort on each other. The only information needed is for the client to know the resource URIs on the server. The interaction between them is only in the form of requests initiated by the client and responses that the server sends to the client in response to requests. The client/server constraint encourages the separation of concerns between the client and the server and allows them to evolve independently.

Stateless

REST services have to be stateless. Each request contains all the information the server needs to perform the request and return a response, regardless of other requests made by the same API user. The server should not require any additional information from previous requests to fulfill the current request. The URI identifies the resource, and the body contains the state of the resource. A stateless service is easy to scale horizontally, allowing additional servers to be added or removed as necessary without worrying about routing subsequent requests to the same server. The servers can be further load-balanced as necessary.

Cache

Response data must be implicitly or explicitly labeled with REST services as cacheable or non-cacheable. The service indicates the duration for which the response is valid. Caching helps improve performance on the client side and scalability on the server side. If the client can access a valid cached reply for a given request, it avoids repeating the same request. Instead, it uses its cached copy. This helps alleviate some of the server's work and thus contributes to scalability and performance.

Note

GET requests should be cacheable by default. Usually, browsers treat all GET requests as cacheable.

POST requests are not cacheable by default but can be made cacheable by adding either an Expires header or a Cache-Control header to the response.

PUT and DELETE are not cacheable at all.

Uniform Interface

A uniform interface is a contract for communication between a client and a server. It is achieved through four sub-constraints:

- **Identification of resources:** As we saw earlier in the chapter, URIs uniquely identify resources. These identifiers are stable and do not change across interactions, even when the resource state changes.
- **Manipulation of resources through representations:** A client manipulates resources by sending new resource representations to the service. The server controls the resource representation and can accept or reject the new resource representation sent by the client.
- **Self-descriptive messages:** REST request and response messages contain all information needed for the service and the client to interpret and handle the message appropriately. The messages are verbose and include the method, protocol, and content type. This enables each message to be independent.
- **Hypermedia as the Engine of Application State (HATEOS):** Hypermedia connects resources and describes their capabilities in machine-readable ways. Hypermedia refers to the hyperlinks, or simply links, that the server can include in the response. Hypermedia is a way for a server to tell a client what HTTP requests the client might want to make in the future.

Layered System

A layered system further builds on the concept of client/server architecture. A layered system indicates that there can be more components than just the client and the server, and each system can have additional layers. These layers should be easy to add, remove, or change. Proxies, load balancers, and so on are examples of additional layers.

Code on Demand

Code on demand is an optional constraint that gives the client flexibility by allowing it to download code. The client can request code from the server, and then the response from the server will contain some code, usually in the form of a script when the response is in HTML format. The client can then execute that code.

REST API Versioning

Versioning is a crucial part of API design. It allows developers to improve an API without breaking the client's applications when new updates are rolled out. Four strategies are commonly employed with API versioning:

- **URI path versioning:** In this strategy, the version number of the API is included in the URL path.
- **Query parameter versioning:** In this strategy, the version number is sent as a query parameter in the URL.
- **Custom headers:** REST APIs are versioned by providing custom headers with the version number included as an attribute. The main difference between this approach and the previous one is that it doesn't clutter the URI with versioning information.
- **Content negotiation:** This strategy allows you to version a single resource representation instead of an entire API, giving you more granular control over versioning. Another advantage of this approach is that it doesn't require you to implement URI routing rules introduced by versioning through the URI path.

Pagination

When a request is made to get a list, it is seldom a good idea to return all resources at once. This is where a pagination mechanism comes into play. There are two popular approaches to pagination:

- Offset-based pagination
- Keyset-based pagination, also known as continuation token or cursor pagination (recommended), a straightforward approach to offset-based pagination is to use the parameters **offset** and **limit**, which are well known from databases.

Example 7-4 shows how query parameters are passed in the URI to get data based on offset and to limit the number of results returned.

Example 7-4 *Pagination: Offset and Limit*

```
http://myhouse.cisco.com/api/room/livingroom/devices?offset=100&limit=15 #
returns the devices between 100-115
```

Usually if the parameters are not specified, the default values are used. Never return all resources. One rule of thumb is to model the limit based on the design of your store retrieval performance.

Example 7-5 shows a URI where no parameters are passed, which results in the default number of results.

Example 7-5 *Pagination: No Parameters Yields the Default*

```
http://myhouse.cisco.com/api/room/livingroom/devices # returns the devices 0
to 200
```

Note that the data returned by the service usually has links to the next and the previous pages, as shown in Example 7-6.

Example 7-6 *Pagination Response Containing Links*

```
GET /devices?offset=100&limit=10
{
  "pagination": {
    "offset": 100,
    "limit": 10,
    "total": 220,
  },
  "device": [
    //...
  ],
  "links": {
    "next": "http://myhouse.cisco.com/devices?offset=110&limit=10",
    "prev": "http://myhouse.cisco.com/devices?offset=90&limit=10"
  }
}
```

Rate Limiting and Monetization

Rate limiting is an essential REST API design method for developers. Rate-limiting techniques are used to increase security, business impact, and efficiency across the board or end-to-end. Let's look at how rate limiting helps each of them:

- **Security:** Allowing unlimited access to your API is like handing over the master key to a house and all the rooms therein. While it's great when people want to use your API and find it useful, open access can decrease value and limit business success. Rate limiting is a critical component of an API's scalability. Processing limits are typically measured in transactions per second (TPS).

  If a user sends too many requests, API rate limiting can throttle client connections instead of disconnecting them immediately. Throttling enables clients to keep using your services while still protecting your API. Finally, keep in mind that there is always a risk of API requests timing out, and the open connections also increase the risk of DDoS attacks. (DDoS stands for distributed denial of service. A DDoS attack consists of a website being flooded by requests during a short period, intending to overwhelm the site and cause it to crash.)

- **Business impact:** One approach to API rate limiting is to offer a free tier and a premium tier, with different limits for each tier. Limits could be regarding sessions or the number of APIs per day or month. There are many factors to consider when deciding what to charge for premium API access. API providers must consider the following when setting up API rate limits:

  - Are requests throttled when they exceed the limit?
  - Do new calls and requests incur additional fees?
  - Do new calls and requests receive a particular error code, and, if so, which one?

- **Efficiency:** Unregulated API requests usually and eventually lead to slow page load times for websites. Not only does this leave customers with an unfavorable opinion, but it can lower your service rankings.

Rate Limiting on the Client Side

As discussed in the previous section, various rate-limiting factors can be deployed on the server side. As a good programming practice, if you are writing client-side code, consider the following:

- Avoid constant polling by using webhooks to trigger updates.
- Cache your data when storing specialized values or rapidly reviewing huge data sets.
- Query with special filters to avoid re-polling unmodified data.
- Download data during off-peak hours.

REST Tools

Understanding and testing REST API architecture when engaging in software development is crucial for any development process. The following sections explore a few of the most commonly used tools in REST API testing and how to use some of their most important features. Based on this information, you will better understand how to determine which one suits a particular development process the best.

Postman

One of the most intuitive and popular HTTP clients is the Postman tool (https://www.getpostman.com/downloads/). It has a straightforward user interface and is very easy to use, even if you're just starting with RESTful APIs. It can handle the following:

- Sending simple GETs and POSTs
- Creating and executing collections (to group requests and run those requests in a predetermined sequence)
- Writing tests (scripting requests with the use of dynamic variables, passing data between requests, and so on)
- Chaining, which allows you to use the output of response as an input to another request
- Generating simple code samples in multiple programming languages
- Importing and executing collections created by the community

Now, let's take a look at several Postman examples. Figure 7-12 shows the user interface of Postman calling an HTTP GET to the Postman Echo server, and Figure 7-13 shows how easy it is to send a POST request using the same interface.

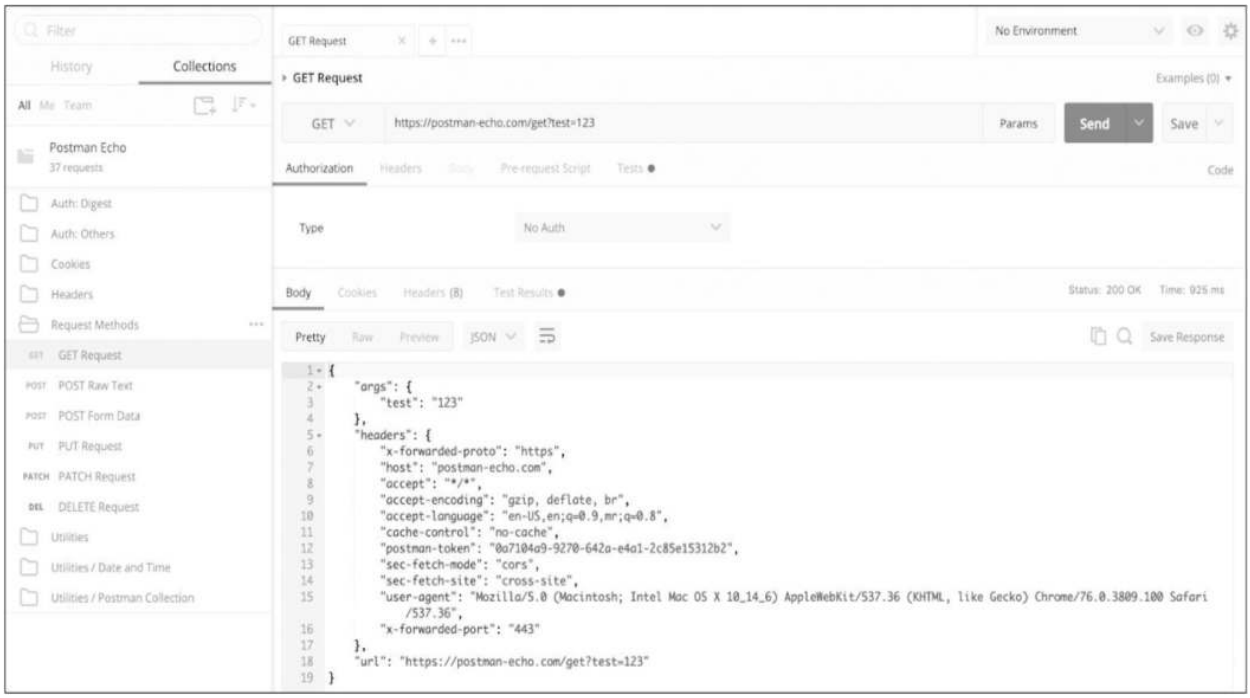


Figure 7-12 *Postman: HTTP GET from the Postman Echo Server*

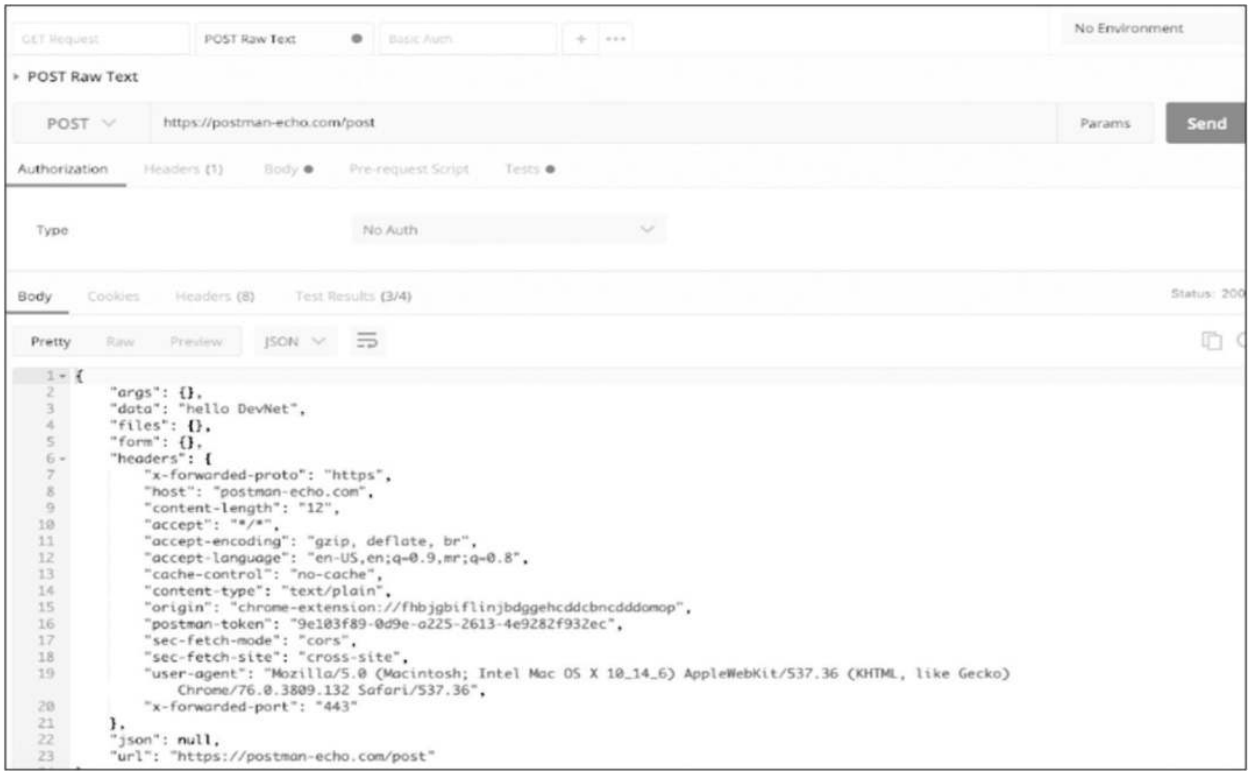


Figure 7-13 *Postman: HTTP POST to the Postman Echo Server*

Figure 7-14 illustrates collections. A collection lets you group individual requests. You can then organize these requests into folders. Figure 7-14 shows the user interface of Postman, with a default collection that interacts with the Postman Echo Server. This interface is a perfect way to learn about various options for sending or getting REST-based information.

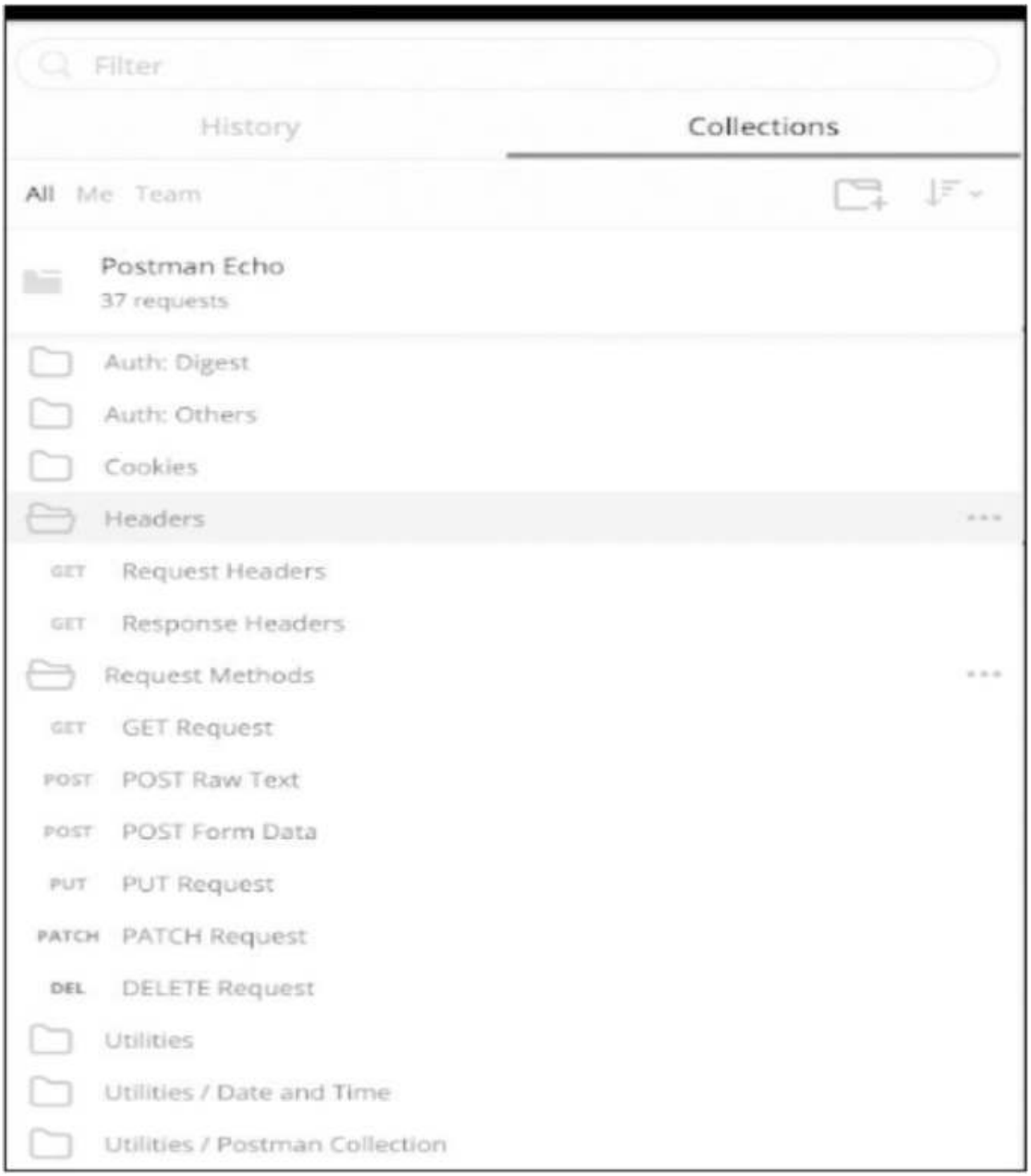


Figure 7-14 *Postman Collection*

It can generate code for any REST API call you try in Postman. After a GET or POST call is made, you can use the Generate Code option and choose the language you prefer. Figure 7-15 shows an example of generating Python code for a simple GET request.

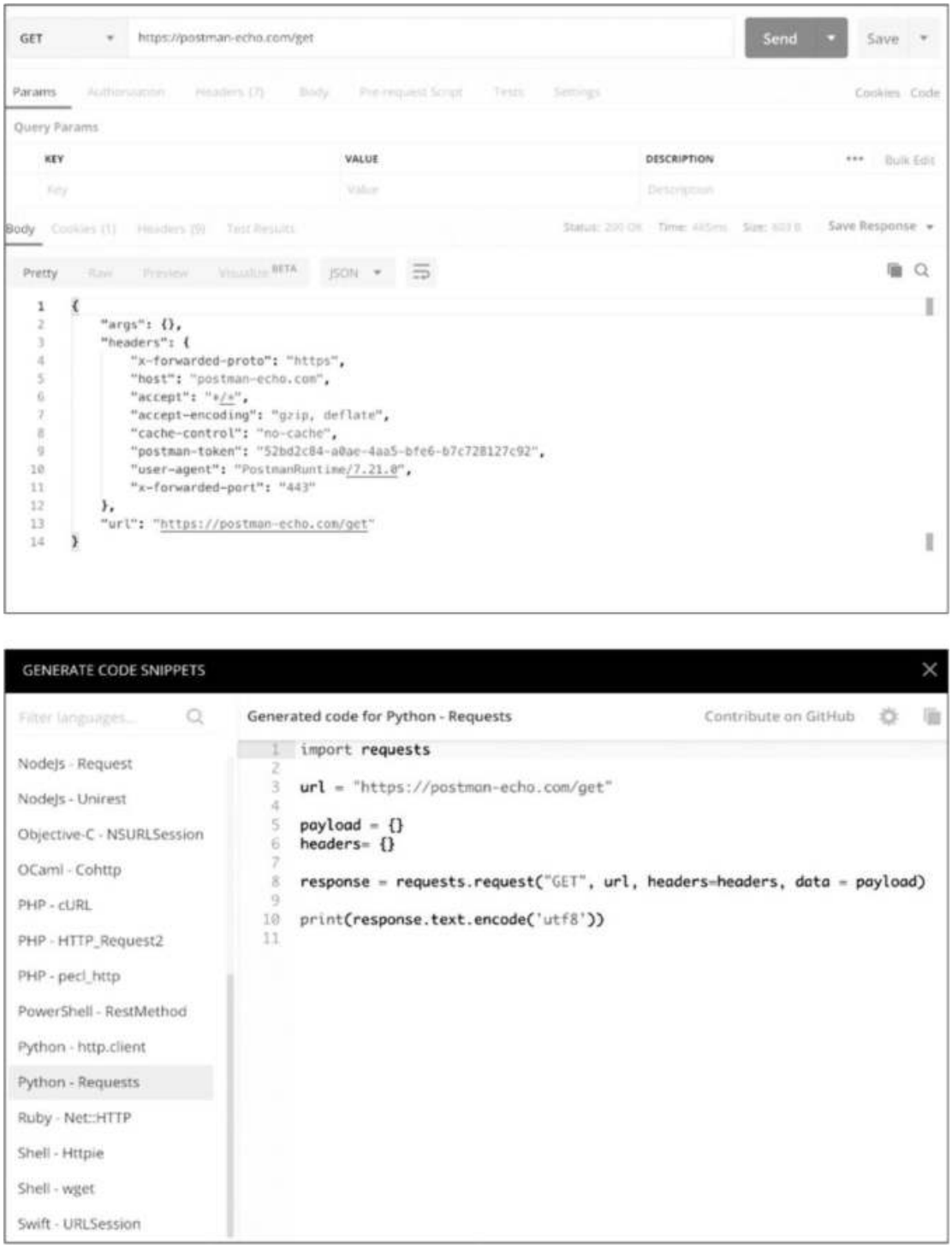


Figure 7-15 *Postman Automatic Code Generation*

Postman has other helpful features, such as environments. An environment is a key/value pair. The key represents the variable's name, allowing you to customize requests; using variables allows you to easily switch between different setups without changing your requests.

Finally, Postman stores a history of past calls so you can quickly reissue a call. Postman even includes some nice touches, such as autocompletion for standard HTTP headers and support for rendering a variety of payloads, including JSON, HTML, and even multipart payloads.

You can find Postman examples at https://learning.postman.com/docs/postman/launching-postman/introduction/.

curl

curl is an extensive command-line tool that can be downloaded from https://curl.haxx.se. Curl can be used on any platform on any hardware today, regardless of what you are running and where the most basic curl commands work.

With curl, you commonly use a couple of command-line options:

- **-d:** This option allows you to pass data to the remote server. You can embed the data in the command or pass the data using a file.
- **-H:** This option allows you to add an HTTP header to the request.
- **-insecure:** This option tells curl to ignore HTTPS certificate validation.
- **-c:** This option stores data received by the server. You can reuse this data in subsequent commands with the -b option.
- **-b:** This option allows you to pass cookie data.
- **-X:** This option allows you to specify the HTTP method, which normally defaults to GET.

Now, let's look at some examples of how to use curl. Example 7-7 shows how to use curl to call a simple GET request.

Example 7-7 *Sample HTTP GET Using curl*

```
$ curl -sD - https://postman-echo.com/get?test=123
HTTP/1.1 200 OK
Content-Type: application/json; charset=utf-8
Date: Tue, 27 Aug 2019 04:59:34 GMT
ETag: W/"ca-42Kz98xXW2nwFREN74xZNS6JeJk"
Server: nginx
set-cookie:
sails.sid=s%3AxZUPHE3Ojk1yts3qrUFqTj_MzBQZZR5n.NrjPkNm0WplJ7%2F%2BX9O7VUTFpKH
pJySLzBytRbnlzYCw; Path=/; HttpOnly
Vary: Accept-Encoding
Content-Length: 202
Connection: keep-alive

{"args":{"test":"123"},"headers":{"x-forwarded-
proto":"https","host":"postman-echo.com","accept":"*/*","user-
agent":"curl/7.54.0","x-forwarded-port":"443"},"url":"https://postman-
echo.com/get?test=123"}
```

Example 7-8 shows how to use curl to call a simple POST request.

Example 7-8 *Sample HTTP POST Using curl*

```
$ curl -sD - -X POST https://postman-echo.com/post -H 'cache-control: no-
cache' -H 'content-type: text/plain' -d 'hello DevNet'
HTTP/1.1 200 OK
Content-Type: application/json; charset=utf-8
Date: Tue, 27 Aug 2019 05:16:58 GMT
ETag: W/"13a-0lMLfkxl7vDVWfb06pyVxdZlaug"
Server: nginx
set-cookie:
sails.sid=s%3AwiFXmSNJpzY0ONduxUCAE8IodwNg9Z2Y.j%2BJ5%2BOmch8XEq8jO1vzH8kjNBi
8ecJij1rGT8D1nBhE; Path=/; HttpOnly
Vary: Accept-Encoding
Content-Length: 314
Connection: keep-alive

{"args":{},"data":"hello DevNet","files":{},"form":{},"headers":{"x-
forwarded-proto":"https","host":"postman-echo.com","content-
length":"12","accept":"*/*","cache-control":"no-cache","content-
type":"text/plain","user-agent":"curl/7.54.0","x-forwarded-
port":"443"},"json":null,"url":"https://postman-echo.com/post"}
```

Example 7-9 shows how to use curl to call a simple GET request with Basic Auth sent via the header.

Example 7-9 *Basic Auth Using curl*

```
$ curl -sD - -X GET https://postman-echo.com/basic-auth -H 'authorization:
Basic cG9zdG1hbjpwYXNzd29yZA==' -H 'cache-control: no-cache'
HTTP/1.1 200 OK
Content-Type: application/json; charset=utf-8
Date: Tue, 27 Aug 2019 05:21:00 GMT
ETag: W/"16-sJz8uwjdDv0wvm7//BYdNw8vMbU"
Server: nginx
set-cookie: sails.sid=s%3A4i3UW5-DQCMpey8Z1Ayrqq0izt4KZR5-
.Bl8QDnt44B690E8J06qyCs8oyCLpUfEsFxLEFTSWSC4; Path=/; HttpOnly
Vary: Accept-Encoding
Content-Length: 22
Connection: keep-alive
{"authenticated":true}
```

HTTPie

HTTPie is a modern, user-friendly, and cross-platform command-line HTTP client written in Python. It is designed to make CLI interaction with web services user-friendly. It's simple HTTP commands enable users to send HTTP requests using intuitive syntax. HTTPie is used primarily for testing, trouble-free debugging, and interacting with HTTP servers, web services, and RESTful APIs. For further information on HTTPie documentation, downloading, and installation, see https://httpie.org/doc:

- HTTPie has an intuitive UI and supports JSON.
- It uses expressive and intuitive command syntax.
- HTTPie allows for syntax highlighting, formatting, and colorized terminal output.
- HTTPie allows you to use HTTPS, proxies, and authentication.
- It provides support for forms and file uploads.
- It provides support for arbitrary request data and headers.
- It enables Wget-like downloads and extensions.

Now, let's take a look at some examples of using HTTPie. Example 7-10 shows how to use HTTPie to call a simple GET request.

Example 7-10 *Sample HTTP GET Using HTTPie*

```
$ http https://postman-echo.com/get?test=123
HTTP/1.1 200 OK
Connection: keep-alive
Content-Encoding: gzip
Content-Length: 179
Content-Type: application/json; charset=utf-8
Date: Tue, 27 Aug 2019 05:27:17 GMT
ETag: W/"ed-mB0Pm0M3ExozL3fgwq7UlH9aozQ"
Server: nginx
Vary: Accept-Encoding
set-cookie: sails.sid=s%3AYCeNAWJG7Kap5wvKPg8HYlZI5SHZoqEf.r7Gi96fe5g7%2FSp0jaJk%2Fa
VRpHZp3Oj5tDxiM8TPZ%2Bpc; Path=/; HttpOnly

{
    "args": {
        "test": "123"
    },
    "headers": {
        "accept": "*/*",
        "accept-encoding": "gzip, deflate",
        "host": "postman-echo.com",
        "user-agent": "HTTPie/1.0.2",
        "x-forwarded-port": "443",
        "x-forwarded-proto": "https"
    },
    "url": "https://postman-echo.com/get?test=123"
}
```

Python Requests

Requests is a Python module that you can use to send HTTP requests. It is an easy-to-use library with many possibilities ranging from passing parameters in URLs to sending custom headers and TLS certificate verification. The Requests library is a handy tool you can use when you programmatically start using any APIs. Here, you will see how to use this library to send simple HTTP requests in Python as a way to illustrate its ease of use.

You can use Requests with Python versions 2.7 and 3.x. Requests is an external module, so it needs to be installed before you can use it. Example 7-11 shows the command you use to install the Requests package for Python.

Example 7-11 *Installing the Requests Package for Python*

```
$ pip3 install requests
```

To add HTTP headers to a request, you can simply pass them in a Python dict to the headers parameter. Similarly, you can send your cookies to a server by using a dict passed to the cookies parameter. Example 7-12 shows a simple Python script that uses the Requests library and does a GET request to the Postman Echo server.

Example 7-12 *Simple HTTP GET Using Python Requests*

```
import requests
url = "https://postman-echo.com/get"
querystring = {"test":"123"}
headers = {}
response = requests.request("GET", url, headers=headers, params=querystring)
print(response.text)
```

Example 7-13 shows a simple Python script that uses the Requests library and makes a POST request to the Postman Echo server. Notice that the headers field is populated with the content type and a new field called payload that sends random text to the service. The response to the request is stored in a response object called response. Everything in the response can be parsed, and further actions can be taken. This example simply prints the values of a few attributes of the response.

Example 7-13 *Simple HTTP POST Using Python Requests*

```
import requests
url = "https://postman-echo.com/post"
payload = "hello DevNet"
headers = {'content-type': "text/plain"}
response = requests.request("POST", url, data=payload, headers=headers)
print(response.text)
```

Example 7-14 shows a simple Python script that uses the Requests library and does a GET request to the Postman Echo server. One difference between Examples 7-11 and 7-14 is related to authentication. With the Requests library, authentication is usually done by passing the 'authorization' keyword along with the type and key.

Example 7-14 *Basic Auth Using Python Requests*

```
import requests
url = "https://postman-echo.com/basic-auth"
headers = {
     'authorization': "Basic cG9zdG1hbjpwYXNzd29yZA=="
}
response = requests.request("GET", url, headers=headers)
print(response.text)
```

REST API Debugging Tools for Developing APIs

As you start playing with RESTful APIs, you are bound to encounter errors. You can use several techniques to determine the nature of a problem. As you saw in Table 7-3, RESTful APIs use several mechanisms to indicate the results of REST calls and errors during processing. You can use these methods to start your debugging journey for a RESTful application. Usually, the error code returned is the biggest hint you can receive. Once you have this information, you can use tools like Postman and Curl to make simple API calls and see the sent and response headers. In addition, other tools that are built into web browsers can allow you to see traces and do other types of debugging.

Most browsers include developer tools like Safari's Web Development Tools, Chrome's DevTools, and Firefox's Developer Tools. Such tools are included with browsers by default, enabling you to inspect API calls quickly. Finally, if you plan on building your test environment or sandbox, you might want to use tools like Simple JSON Server (an open-source server you can clone and run in your environment for playing with and learning about RESTful APIs).

Exam Preparation Tasks

As mentioned in the section "How to Use This Book" in the Introduction, you have a couple of choices for exam preparation: the exercises here, Chapter 19, "Final Preparation," and the exam simulation questions in the Pearson Test Prep Software Online.

Review All Key Topics

Review the most important topics in this chapter, noted with the Key Topic icon in the outer margin of the page. Table 7-5 lists these key topics and the page number on which each is found.

Table 7-5 *Key Topics for Chapter 7*

Key Topic Element	Description	Page
List	The elements of a URL	
Table 7-2	Request Methods	
Table 7-4	HTTP Status Codes	
Paragraphs	Data formats and XML, YAML, and JSON data	
Section	REST Constraints	
Paragraph	Pagination	
Section	REST Tools	

Define Key Terms

Define the following key terms from this chapter and check your answers in the glossary:

API
REST
CRUD
YAML
JSON
webhook

Chapter 8. Cisco Enterprise Networking Management Platforms and APIs

This chapter covers the following topics:

- **What Is an SDK?:** This section covers what an SDK is and what it is used for.
- **Cisco Meraki:** This section covers the Cisco Meraki platform and the REST APIs it exposes.
- **Cisco Catalyst Center:** This section covers Cisco Catalyst Center and the REST APIs that it publicly exposes.
- **Cisco Catalyst SD-WAN:** This section covers Cisco Catalyst SD-WAN and the REST APIs exposed through Cisco Manager.

In Chapter 7, "RESTful API Requests and Responses," you learned about REST API concepts. This chapter begins exploring software development kits (SDKs) as well as Cisco enterprise networking products, their APIs, and the public SDKs that come with them. In particular, this chapter explores the Cisco Meraki, Cisco Catalyst Center, and Cisco Catalyst SD-WAN platforms and the REST APIs they expose. This chapter provides a short introduction to each of these solutions and shows authentication and authorization API calls for each platform. This chapter also covers basic API calls, such as for obtaining a list of devices and client health status. API tools such as curl and Postman are used throughout the chapter. Python SDKs and scripts are also explored as an introduction to network programmability and automation.

"Do I Know This Already?" Quiz

The "Do I Know This Already?" quiz allows you to assess whether you should read this entire chapter thoroughly or jump to the "Exam Preparation Tasks" section. If you are in doubt about your answers to these questions or your own assessment of your knowledge of the topics, read the entire chapter. Table 8-1 lists the major headings in this chapter and their corresponding "Do I Know This Already?" quiz questions. You can find the answers in Appendix A, "Answers to the 'Do I Know This Already?' Quiz Questions."

Table 8-1 *"Do I Know This Already?" Section-to-Question Mapping*

Foundation Topics Section	Questions
What Is an SDK?	1–2
Cisco Meraki	3–5
Cisco Catalyst Center	6–8
Cisco Catalyst SD-WAN	9–10

Caution

The goal of self-assessment is to gauge your mastery of the topics in this chapter. If you do not know the answer to a question or are only partially sure of the answer, you should mark that question as wrong for purposes of self-assessment. Giving yourself credit for an answer that you correctly guess skews your self-assessment results and might provide you with a false sense of security.

1. What are some of the features of a good SDK? (Choose three.)
   - **a.** Is easy to use
   - **b.** Is well documented
   - **c.** Integrates well with other SDKs
   - **d.** Impacts hardware resources

2. What are the advantages of using an SDK? (Choose two.)
   a. Quicker integration
   b. Faster development
   c. Advanced customization
   d. Error handling
3. What APIs does the Cisco Meraki platform provide to developers? (Choose two.)
   a. Captive Portal API
   b. Scanning API
   c. Access Point API
   d. Infrastructure API
4. What is the name of the Cisco Meraki Dashboard API authentication header?
   a. X-Cisco-Meraki-API-Key
   b. X-Cisco-Meraki-Token
   c. X-Cisco-Meraki-Session-key
   d. Bearer
5. What is the base URL for the Cisco Meraki Dashboard API?
   a. https://api.meraki.com/api/v1
   b. https://api.meraki.com/v1/api
   c. https://api.meraki.cisco.com/api/v0
   d. https://api.meraki.cisco.com/v1/api
6. What type of authentication do the Cisco Catalyst Center platform APIs use?
   a. No auth
   b. API key
   c. Basic auth
   d. Hash-based message authentication
7. When you specify the **timestamp** parameter with the Cisco Catalyst Center APIs, what format should the time be in?
   a. UNIX time
   b. OS/2
   c. OpenVMS
   d. SYSTEMTIME
8. Which component of the Cisco Catalyst SD-WAN fabric exposes a public REST API interface?
   a. Controller
   b. Validator
   c. Manager
   d. Edge
9. When initially authenticating to the Cisco Catalyst SD-WAN REST API, how are the username and password encoded?
   a. application/postscript
   b. application/xml
   c. application/json
   d. application/x-www-form-urlencoded

Foundation Topics

What Is an SDK?

A *software development kit (SDK),* or devkit, is a set of software development tools that developers can use to create software or applications for a certain platform, operating system, computer system, or device. An SDK typically contains a set of libraries,

APIs, documentation, tools, sample code, and processes that make it easier for developers to integrate, develop, and extend the platform. An SDK is created for a specific programming language, and it is very common to have the same functionality exposed through SDKs in different programming languages.

Chapter 6, "Application Programming Interfaces (APIs)," describes what an API is, and Chapter 7 covers the most common API framework these days, the REST API framework. As a quick reminder, an application programming interface (API) is, as the name implies, a programming interface that implements a set of rules developers and other software systems can use to interact with a specific application.

As you will see in this chapter and throughout this book, where there is an API, there is usually also an SDK. Software developers can, of course, implement and spend time on developing code to interact with an API (for example, building their own Python classes and methods to authenticate, get data from the API, or create new objects in the API), or they can take advantage of the SDK, which makes all these objects already available.

Besides offering libraries, tools, documentation, and sample code, some SDKs also offer their own integrated development environments (IDEs). For example, the SDKs available for mobile application development on Google Android and Apple iOS also make an IDE available to give developers a complete solution to create, test, debug, and troubleshoot their applications.

A good SDK has these qualities:

- Is easy to use
- Is well documented
- Has value-added functionality
- Integrates well with other SDKs
- Has minimal impact on hardware resources

To be used by developers, an SDK should be easy to use and follow best practices for software development in the programming language for which the SDK was developed. For example, for Python development, there are *Python Enhancement Proposals (PEPs),* which are documents that provide guidance and spell out best practices for how Python code should be organized, packaged, released, deprecated, and so on. PEP8 is a popular standard for styling Python code and is extensively used in the developer community. Documenting the SDK inline as well as having external documentation is critical to developer adoption and the overall quality of the SDK. Having good, up-to-date documentation of the SDK makes the adoption and understanding of the code and how to use it much easier. A good SDK also adds value by saving development time and providing useful features. Integrating with other SDKs and development tools should be easy and scalable, and the code should be optimized for minimal hardware resource utilization as well as execution time.

SDKs provide the following advantages:

- Quicker integration
- Faster and more efficient development
- Brand control
- Increased security
- Metrics

As mentioned previously, there are significant development time savings when adopting SDKs, because the functionality and features provided by an SDK don't have to be developed in house. This time savings leads to quicker integration with the API and quicker time to market. In addition, brand control can be enforced with the SDK. The look and feel of applications developed using the SDK can be uniform and in line with the overall brand design. For example, applications developed for iOS using the Apple SDK have a familiar and uniform look and feel because they use the building blocks provided by the SDK. Application security best practices can be enforced through SDKs. When you develop using a security-conscious SDK, the applications developed have the SDK's security features integrated automatically. Sets of metrics and logging information can be included with an SDK to provide better insights into how the SDK is being used and for troubleshooting and performance tweaking.

It is critical to ensure a great experience for all developers when interacting with an API. In addition, offering a great SDK with an API is mandatory for success.

Cisco has been developing applications and software since its inception. As the requirements for integrations with other applications and systems have grown, APIs have been developed to make it easier for developers and integrators to create and

develop their own solutions and integrations. Throughout the years, software architectures have evolved, and currently all Cisco solutions provide some type of API. As mentioned earlier, where there is an API, there is usually also an SDK.

The starting point in exploring all the SDKs that Cisco has to offer is https://developer.cisco.com. As you will see in the following sections of this chapter and throughout this book, several SDKs developed by Cisco and third parties take advantage of the APIs that currently exist with all Cisco products.

Cisco Meraki

Meraki became part of Cisco following its acquisition in 2012. The Meraki portfolio is large, comprising wireless, switching, security, and video surveillance products. The differentiating factor for Meraki, compared to similar products from Cisco and other vendors, is that management is cloud based. You can explore all the current Cisco Meraki products and offerings at https://meraki.cisco.com.

From a programmability perspective, the Meraki cloud platform provides several APIs:

- Captive Portal API
- Scanning API
- MV Sense Camera API
- Dashboard API

The Cisco Meraki cloud platform also provides webhooks, which offer a powerful and lightweight way to subscribe to alerts sent from the Meraki cloud when an event occurs. (For more information about webhooks, see Chapter 7.) A Meraki alert includes a JSON-formatted message that can be configured to be sent to a unique URL, where it can be further processed, stored, and acted upon to enable powerful automation workflows and use cases.

The Captive Portal API extends the power of the built-in Meraki splash page functionality by providing complete control of the content and authentication process that a user interacts with when connecting to a Meraki wireless network. This means Meraki network administrators can completely customize the portal, including the onboarding experience for clients connecting to the network, how the web page looks and feels, and the authentication and billing processes.

The Scanning API takes advantage of Meraki smart devices equipped with wireless and *Bluetooth Low Energy (BLE)* functionality to provide location analytics and report on user behavior. This capability can be especially useful in retail, healthcare, and enterprise environments, where business intelligence and information can be extracted about trends and user engagement and behavior. The Scanning API delivers data in real time and can be used to detect Wi-Fi and BLE devices and clients. The data is exported to a specified destination server through an HTTP POST of JSON documents. At the destination server, this data can then be further processed, and applications can be built on top of the received data. Taking into consideration the physical placement of the access points on the floor map, the Meraki cloud can estimate the location of the clients connected to the network. The geolocation coordinates of this data vary based on a number of factors and should be considered as a best-effort estimate.

The MV Sense Camera API takes advantage of the powerful onboard processor and a unique architecture to run machine learning workloads at the edge. Through the MV Sense API, object detection, classification, and tracking are exposed and become available for application integration. You can, for example, extract business insight from video feeds at the edge without the high cost of compute infrastructure that is typically needed with computer imagining and video analytics. Both REST and MQTT API endpoints are provided, and information is available in a request or subscribe model. *MQ Telemetry Transport (MQTT)* is a client/server publish/subscribe messaging transport protocol. It is lightweight, simple, open, and easy to implement. MQTT is ideal for use in constrained environments such as Internet of Things (IoT) and machine-to-machine communication where a small code footprint is required.

The Meraki APIs covered so far are mostly used to extract data from the cloud platform and build integrations and applications with that data. The Dashboard API, covered next, provides endpoints and resources for configuration, management, and monitoring automation of the Meraki cloud platform. The Dashboard API is meant to be open ended and can be used for many purposes and use cases. Some of the most common use cases for the Dashboard API are as follows:

- Provisioning new organizations, administrators, networks, devices, and more
- Configuring networks at scale
- Onboarding and decommissioning of clients
- Building custom dashboards and applications

To get access to the Dashboard API, you need to create an API key. Begin by logging in to the Cisco Meraki dashboard at https://dashboard.meraki.com using your favorite web browser and navigating to **Organization > Configure > API & Webhooks**. From there, click on the API keys and access tab, and you will have the option to generate a new API key. The API

key you generate is associated with your account. You can generate, revoke, and regenerate your API key here in the API & Webhooks section or in your profile. Make sure you copy and store your API key in a safe place, because whoever has this key can impersonate you and get access through the Dashboard API to all the information your account has access to. For security reasons, the API key is not stored in plaintext in your profile, so if you lose the key, you will have to revoke the old one and generate a new one. If you believe that your API key has been compromised, you can generate a new one to automatically revoke the existing API key.

Every Dashboard API request must specify an API key within the request header. If a missing or incorrect API key is specified, the API returns a 404 HTTP error message. The key for the authentication request header is X-Cisco-Meraki-API-Key, and the value is the API key you obtained previously.

To mitigate abuse and denial-of-service (DoS) attacks, the Cisco Meraki Dashboard API is limited to 10 API calls per second per organization. If the rate limit has been exceeded, an error message with HTTP status code 429 is returned. The rate-limiting technique that the Dashboard API implements is based on the token bucket model. The token bucket is an algorithm used to check that the data that is transmitted in a certain amount of time complies with set limits for bandwidth and burstiness. Based on this model, if the number of API requests crosses the set threshold for a certain amount of time, you have to wait a set amount of time until you can make another request to the API. The time you have to wait depends on how many more requests you have performed above the allowed limit; the more requests you have performed, the more time you have to wait.

The Cisco Meraki Dashboard API uses the base URL https://api.meraki.com/api/v1. Keep in mind that the API will evolve, and different versions will likely be available in the future. Always check the API documentation for the latest information on all Cisco APIs, including the Meraki APIs, at https://developer.cisco.com.

Most Dashboard API calls require either the organization ID or the network ID as part of the endpoint. (You will see later in this chapter how to obtain these IDs and how to make Dashboard API calls.) When you have these IDs, you can build and make more advanced calls to collect data, create and update new resources, and configure and make changes to the network. Remember that all API calls require an API key.

If your Meraki dashboard contains a large number of organizations, networks, and devices, you might have to consider pagination when making API calls. Recall from Chapter 7 that pagination is used when the data returned from an API call is too large and needs to be limited to a subset of the results. The Meraki Dashboard API supports three special query parameters for pagination:

- **perPage:** The number of entries to be returned in the current request
- **startingAfter:** A value used to indicate that the returned data will start immediately after this value
- **endingBefore:** A value used to indicate that the returned data will end immediately before this value

While the types of the **startingAfter** and **endingBefore** values differ based on API endpoints, they generally are either timestamps specifying windows in time for which the data should be returned or integer values specifying IDs and ranges of IDs.

The Dashboard API also supports action batches, which make it possible to submit multiple configuration requests in a single transaction and are ideal for initial provisioning of a large number of devices or performing large configuration changes throughout the whole network. Action batches also provide a mechanism to avoid hitting the rate limitations implemented in the API for high-scale configuration changes, because you can implement all the changes with one or a small number of transactions instead of a large number of individual API requests. Action batch transactions can be run either synchronously, waiting for the API call to return before continuing, or asynchronously, in which case the API call does not wait for a return as the call is placed in a queue for processing. (In Chapter 7 you saw the advantages and disadvantages of both synchronous and asynchronous APIs.) With action batches, you can be confident that all the updates contained in the transaction were submitted successfully before being committed because batches are run in an atomic fashion: all or nothing.

After you have generated the API key, and saved it in a safe place, you are ready to interact with the API. For the rest of this chapter, we recommend using the Meraki Small Business and Enterprise Sandbox, which you can find at https://developer.cisco.com/sandbox. Follow the steps in the sandbox instructions to add your individual administrator account to the sandbox.

At this point, you need to obtain the organization ID for this account. As you saw in Chapter 7, you can interact with an API in several ways: You can use tools like curl and Postman, or you can interact with the API through programming languages and the libraries that they provide. In this case, you will use curl and Postman to get the organization ID for the Cisco DevNet Sandbox account and then the Cisco Meraki Python SDK.

As mentioned earlier, the base URL for the Dashboard API is https://api.meraki.com/api/v1. To get the organizations for the account with the API key you created following the sandbox instructions, you have to append the /organizations resource to the base URL. The resulting endpoint becomes https://api.meraki.com/api/v1/organizations. You also need to include the X-Cisco-Meraki-API-Key header for authentication purposes. This header will contain the API key you generated. The **curl** command should look as follows in this case:

```
curl -I -X GET /

   --url 'https://api.meraki.com/api/v1/organizations' /

   -H 'X-Cisco-Meraki-API-Key:6bec40cf957de430a6f1f2baa056b99a4fac9ea0'
```

The response should look as shown in Example 8-1.

Example 8-1 *Headers of the GET Organizations REST API Call*

```
HTTP/2 200
date: Thu, 08 Jan 2026 01:10:09 GMT
content-type: application/json; charset=utf-8
vary: Accept-Encoding
x-frame-options: sameorigin
x-xss-protection: 1; mode=block
x-xss-protection: 1; mode=block
x-content-type-options: nosniff
x-robots-tag: none
cache-control: no-cache, no-store, max-age=0, must-revalidate
pragma: no-cache
expires: Fri, 01 Jan 1990 00:00:00 GMT
link:
<https://api.meraki.com/api/v1/organizations?perPage=9000&startingAfter=0>;
rel=first,
<https://api.meraki.com/api/v1/organizations?endingBefore=9223372036854775807
&perPage=9000>; rel=last
x-request-id: 8e2f859e139ca490158161ec57654874
x-runtime: 0.043190
strict-transport-security: max-age=31536000; includeSubDomains
cf-cache-status: DYNAMIC
server: cloudflare
cf-ray: 9ba7d044dbcfa709-SJC
```

You can see in Example 8-1 that the response code for the request is 200. This code indicates that the API call has completed successfully and your API key is valid. When you specify the **-I** option for curl, only the headers of the response are displayed to the user. You can run the **curl** command again with the same resource (https://api.meraki.com/api/v1/organizations), remove the **-I** flag, and add an Accept header to specify that the response to the call should be in JSON format. The command should look like this:

```
curl -X GET /
   --url 'https://api.meraki.com/api/v1/organizations' /
    -H 'X-Cisco-Meraki-API-Key: 15da0c6ffff295f16267f88f98694cf29a86ed87'/
    -H 'Accept: application/json'
```

The response in this case contains the ID of the DevNet Sandbox organization in JSON format, as shown in Example 8-2.

Example 8-2 *List of All Organizations Attached to a Certain Meraki User Account*

```
[
  {
    "id": "669910444571366839",
    "name": "Devnet Mqzx2KNBledI",
    "url":
"https://n190.dashboard.meraki.com/o/OuNsPc-c/manage/organization/overview",
    "samlConsumerUrls": [
      "https://n190.dashboard.meraki.com/saml/login/OuNsPc-c/E1TdPakLUkqc"
    ],
    "samlConsumerUrl":
"https://n190.dashboard.meraki.com/saml/login/OuNsPc-c/E1TdPakLUkqc",
    "api": {
      "enabled": true
    },
    "licensing": {
      "model": "co-term"
    },
    "cloud": {
      "region": {
        "name": "North America",
        "host": {
          "name": "United States"
        }
      }
    },
    "management": {
      "details": []
    }
  }
]
```


Now let's look at how to obtain the organization ID for the Cisco DevNet Sandbox Meraki account by using Postman. As mentioned in Chapter 7, Postman is a popular tool used to explore APIs and create custom requests; it has extensive built-in support for different authentication mechanisms, headers, parameters, collections, environments, and so on. Figure 8-1 shows the Postman client interface with all the fields (the method of the call, the resource URL, and the headers) populated so that the Cisco Meraki REST API returns all the organizations to which this account has access.

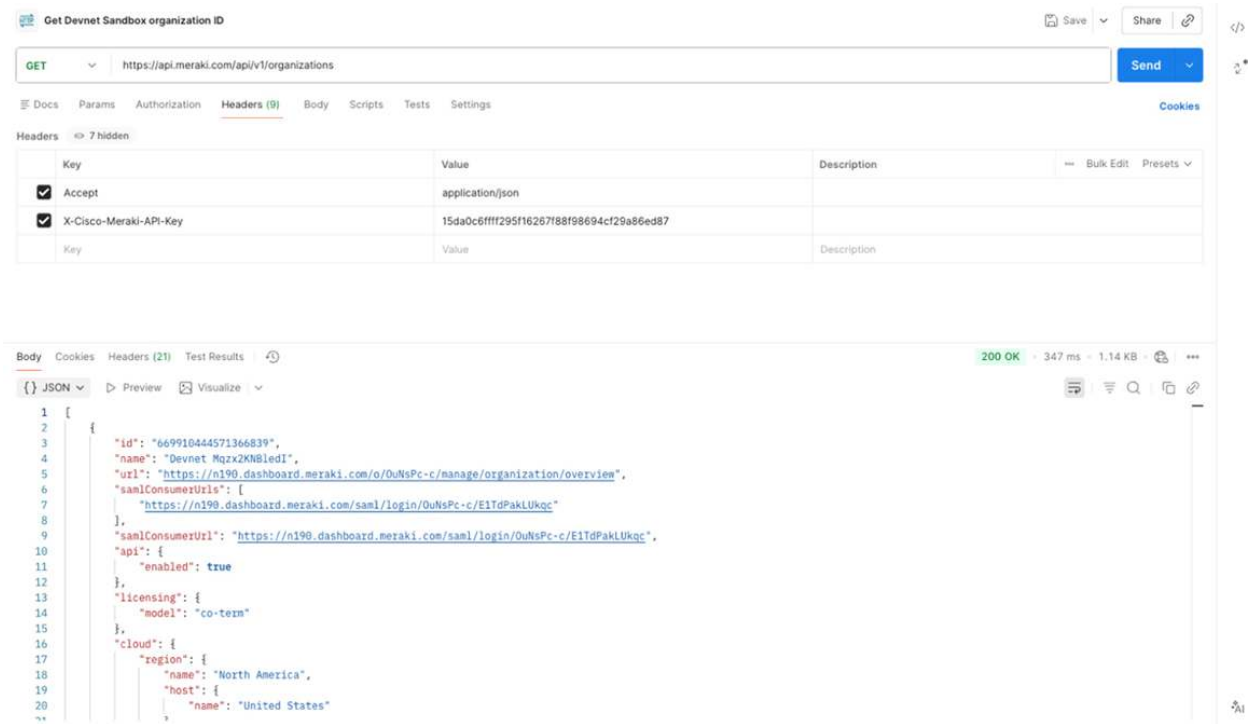


Figure 8-1 *GET Organizations REST API Call in Postman*

In the body of the response, you see the same JSON-formatted output as before, with the same organization ID for the DevNet Sandbox account.

Let's explore the Meraki Dashboard API further and obtain the networks associated with the DevNet Sandbox organization. If you look up the API documentation at https://developer.cisco.com/meraki/api-v1/, you see that to obtain the networks associated with a specific organization, you need to do a GET request to https://api.meraki.com/api/v1/organizations/{organizationId}/networks, where {organizationId} is the ID you obtained previously, in your first interaction with the Dashboard API. You can modify the API endpoint with this information to use the following **curl** command:

```
curl -X GET /

   --url

   'https://api.meraki.com/api/v1/organizations/669910444571366839/networks'/

   -H 'X-Cisco-Meraki-API-Key: 15da0c6ffff295f16267f88f98694cf29a86ed87'/

   -H 'Accept: application/json'
```

The response from the API should contain a list of all the networks that are part of the DevNet Sandbox organization and should look similar to the output in Example 8-3.

Example 8-3 *List of All the Networks in a Specific Organization*

```
[
  {
    "id": "L_669910444571380911",
    "organizationId": "669910444571366839",
    "name": "branch office",
    "productTypes": [
      "appliance",
      "camera",
      "cellularGateway",
      "sensor",
      "switch",
      "wireless"
    ],
    "timeZone": "America/Los_Angeles",
    "tags": [],
    "enrollmentString": null,
    "url":
"https://n190.dashboard.meraki.com/branch-office-ap/n/2qypib-c/manage/clients
",
    "notes": null,
    "details": null,
    "isBoundToConfigTemplate": false,
    "isVirtual": false
  }
]
```

The output in Example 8-3 shows all the networks that are part of the DevNet Sandbox organization with an ID of 669910444571366839. In this case, there is only one network, called branch office. You should make a note of the network ID returned by the API because you will need it in the next step in your exploration of the Meraki Dashboard API.

Now you can try to get the same information—a list of all the networks that are part of the DevNet Sandbox organization—by using Postman. In Postman, you can specify the API endpoint in the same way as you did for the **curl** command: https://api.meraki.com/api/v1/organizations/669910444571366839/networks. You must make sure to specify the GET method, the X-Cisco-Meraki-API-Key header for authentication, and the Accept header, in which you specify that you would like the response from the API to be in JSON format. Figure 8-2 shows the Postman client interface with all the information needed to obtain a list of all the networks that belong to the organization with ID 669910444571366839.

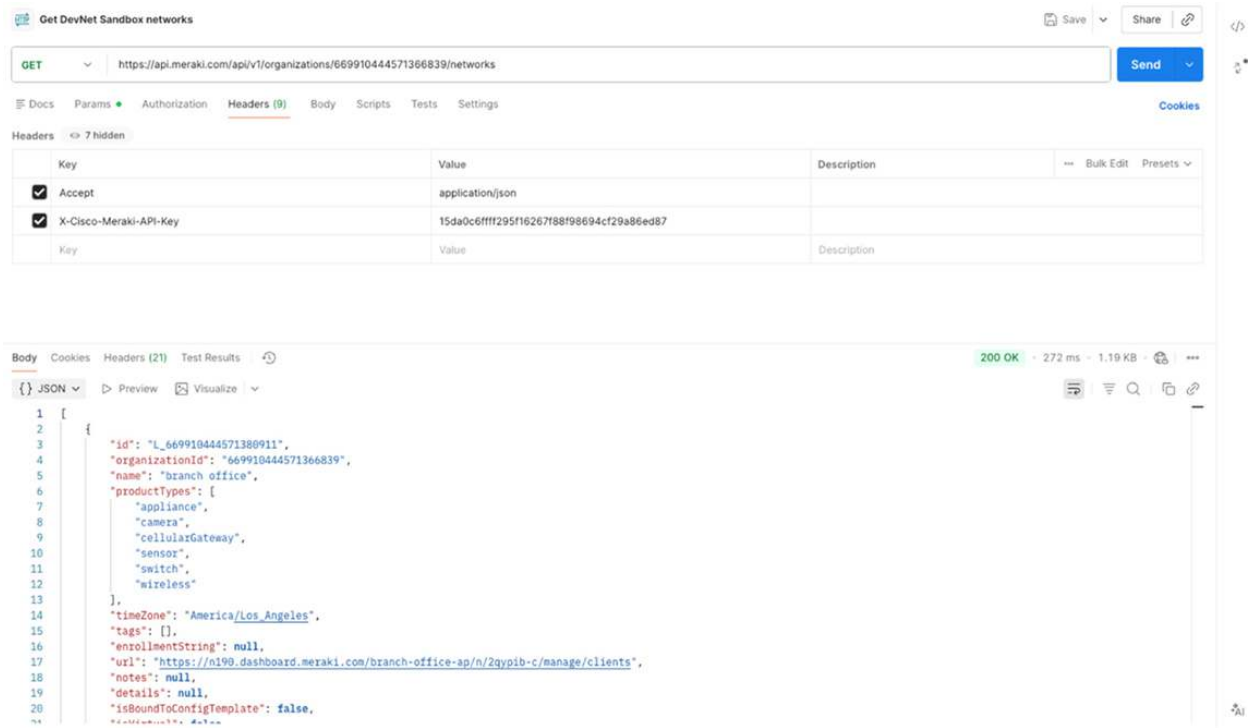


Figure 8-2 *GET Networks REST API Call in Postman*

The body of the response contains exactly the same information as before: one network named branch office with an ID of L_669910444571380911.

Next, you can obtain a list of all devices that are part of the network that has the name “branch office” and ID L_669910444571380911. As in the previous steps, you start by checking the API documentation to find the API endpoint that will return this data to you. The API resource that contains the information you are looking for is /networks/{networkId}/devices, because you can search from the API documentation at the following link: https://developer.cisco.com/meraki/api-v1/. You add the base URL for the DevNet Sandbox account—in this case, https://api.meraki.com/api/v1—and populate {networkId} with the value you obtained in the previous step. When you combine all this information, the endpoint that will return the information you are seeking is https://api.meraki.com/api/v1/networks/L_669910444571380911/devices. The **curl** command in this case is as follows:

```
curl -X GET /

   --url 'https://api.meraki.com/api/v1/networks/L_669910444571380911/devices' /

   -H 'X-Cisco-Meraki-API-Key: 15da0c6ffff295f16267f88f98694cf29a86ed87'/

   -H 'Accept: application/json'
```

The response from the API should be similar to the one in Example 8-4.

Example 8-4 *List of All the Devices in a Specific Network*

```
[
  {
    "lat": 37.4180951010362,
    "lng": -122.098531723022,
    "address": "",
    "serial": "XXXX-YYYY-ZZZZ",
    "mac": "4c:c8:a1:02:00:01",
    "wan1Ip": null,
    "wan2Ip": null,
    "url":
 "https://n190.dashboard.meraki.com/branch-office-ap/n/2qypib-c/manage/nodes/n
 ew_list/84424578433051",
    "networkId": "L_669910444571380911",
    "tags": [],
    "name": null,
    "details": [],
    "model": "MX100",
    "firmware": "Not running configured version",
    "floorPlanId": null
  },
  {
    "lat": 37.4180951010362,
    "lng": -122.098531723022,
    "address": "",
    "serial": "YYYY-XXXX-ZZZZ",
    "mac": "4c:c8:a1:0c:01:00",
    "lanIp": null,
    "url":
 "https://n190.dashboard.meraki.com/branch-office-wi/n/v0DWBa-c/manage/nodes/n
 ew_list/84424579088708",
    "networkId": "L_669910444571380911",
    "tags": [],
    "name": null,
    "details": [],
    "model": "MR52",
    "firmware": "Not running configured version",
    "floorPlanId": null
  },
  {
    "lat": 37.4180951010362,
    "lng": -122.098531723022,
    "address": "",
    "serial": "ZZZZ-XXXX-YYYY",
    "mac": "4c:c8:a1:0f:01:00",
    "lanIp": null,
    "url":
 "https://n190.dashboard.meraki.com/branch-office-ca/n/tNG-Sc-c/manage/nodes/n
 ew_list/84424579285302",
    "networkId": "L_669910444571380911",
    "tags": [],
    "name": null,
    "details": [],
    "model": "MV12W",
    "firmware": "Not running configured version",
    "floorPlanId": null,
    "wirelessMac": "4c:c8:a1:0f:01:37"
  }
]
```

Notice from the API response that this network has three devices: an MX100 security appliance, an MR52 access point, and an MV12W camera. The output also includes geographic coordinates, MAC addresses, serial numbers, tags, model numbers, and other information. The same information is available in the Meraki GUI dashboard.

Following the process used so far, you can obtain the same information but this time using Postman. The same API endpoint is used in Postman: https://api.meraki.com/api/v1/networks/L_669910444571380911/devices. You populate the two headers Accept and X-Cisco-Meraki-API-Key with their respective values, select the GET method, and click the Send button. If everything goes well, the response code should be 200 OK, and the body of the response should contain exactly the same information found using curl. Figure 8-3 shows the Postman client interface with all the headers and fields needed to get a list of all devices that are part of the network with ID L_669910444571380911.

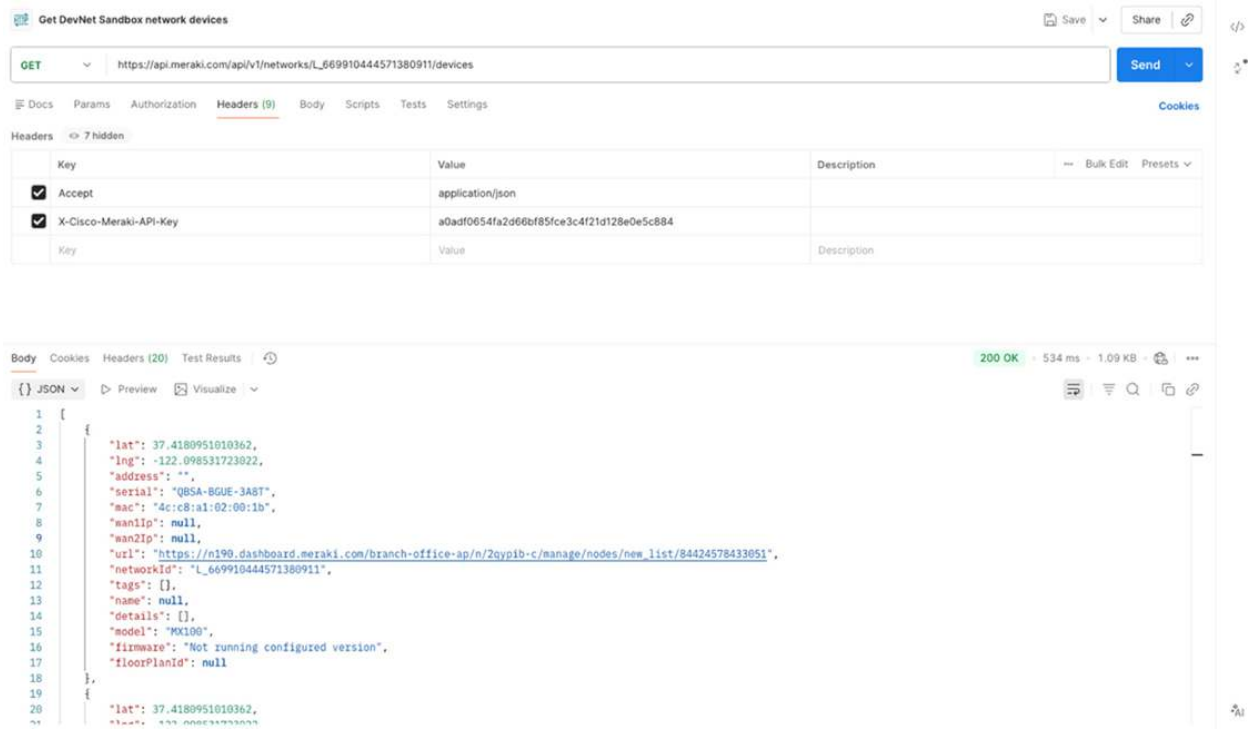


Figure 8-3 *GET Devices REST API Call in Postman*

So far, you have explored the Meraki Dashboard API using curl and Postman. You first obtained the organization ID of the DevNet Sandbox and then, based on that ID, you obtained the networks that are part of the organization. Then you used the only network ID you obtained to find all the devices that are part of that specific network. This is, of course, just a subset of all the capabilities of the Meraki Dashboard API, and we leave it as an exercise for you to explore in more depth all the capabilities and functionalities of the API.

As a final step in this section, let's look at the Meraki Python SDK. As of this writing, there is one Meraki SDK written in Python. The Meraki Python SDK used in this book is version 2.1.0; it was developed for Python 3.7 and higher and implements a complete set of classes, methods, and functions to simplify how users interact with the Dashboard API in Python.

To get access to the SDK, you need to install the meraki module. As a best practice, you should always use virtual environments with all Python projects. Once a virtual environment is activated, you can run **pip install meraki** to get the latest version of the SDK. In this section, you follow the same three steps you have followed in other examples in this chapter: Get the organization ID for the DevNet Sandbox account, get a list of all the networks that are part of this organization, and get all the devices associated with the DevNet Sandbox network. The Python 3 code to accomplish these three tasks might look as shown in Example 8-5.

You need to import the meraki module. You use the DashboardAPI class to create an API client by passing the API key as a parameter and creating an instance of this class called dashboard.

Example 8-5 *Python Script That Uses Meraki SDK*

```
#! /usr/bin/env python
import meraki

# Your Cisco Meraki API key
X_CISCO_MERAKI_API_KEY = "15da0c6ffff295f16267f88f98694cf29a86ed87"

# Instantiate a Meraki dashboard API session
dashboard = meraki.DashboardAPI(api_key=X_CISCO_MERAKI_API_KEY, print_console=False)

# Get list of organizations to which API key has access
organizations = dashboard.organizations.getOrganizations()

# Iterate through list of orgs
for org in organizations:
    print('Org ID: {} and Org Name: {}'.format(org["id"], org["name"]))
    org_id = org["id"]

    # Get list of networks in organization
    nets = dashboard.organizations.getOrganizationNetworks(org_id)

    # Iterate through list of orgs
    for net in nets:
        print('Network ID: {0:20s}, Name: {1:12s}, Tags: {2:<10s}'.format(\
            net["id"], net["name"], str(net["tags"])))
        net_id = net["id"]

        #Get list of all the devices that are part of the network
        devices = dashboard.networks.getNetworkDevices(net_id)
        # Iterate through list of devices
        for device in devices:
            print('Device Model: {0:9s},Serial: {1:14s},MAC: {2:17},
  Firmware:{3:12s}'\
                .format(device["model"], device["serial"], device["mac"], \
                      device["firmware"]))
```

After you instantiate the Meraki class, you get an API client object that provides access to all the methods of the class. You can find the documentation for the Python SDK at https://developer.cisco.com/meraki/api-v1/python/#guides/python-sdk-quick-start. Unlike with the curl and Postman examples earlier in this chapter, you do not need to determine the exact API resource that will return the information you are interested in; however, you do need to know how the Meraki API class is organized and what methods are available. There's a consistent one-to-one mapping between the Dashboard API and the Python SDK, so when you are familiar with one of them, you should find the other one easy to understand. There is also no need to pass the API key through the X-Cisco-Meraki-API-Key header. This is all automatically handled by the SDK.

Obtaining the organization ID for the DevNet Sandbox account is as easy as invoking the **organizations.getOrganizations()** method of the API client class. The **orgs** variable in Example 8-5 contains a list of all the organizations that are part of the DevNet Sandbox. Next, you iterate within a **for** loop through all these organizations and display the organization ID and the organization name to the console.

Next, you save the organization ID into a variable named org_id. You still use the Meraki API client instance, but in this case, you invoke the **organizations.getOrganizationNetworks()** method. Just as in the case of the REST API calls with curl and Postman earlier in this section, where you had to specify the organization ID when building the endpoint to obtain the list of networks, the **getOrganizationNetworks()** method takes as input the organization ID value that is stored in org_id. The **nets** variable stores the output of the API call. In another iterative loop, information about each network is displayed to the console. The network ID is also saved in the **net_id** variable.

Finally, you get the list of devices that are part of the network with a net_id network ID. In this case, you use the **networks.getNetworkDevices()** method of the Meraki API client instance and store the result in the **devices** variable. You iterate over the **devices** variable and, for each device in the list, extract and print to the console the device model, the serial number, the MAC address, and the firmware version.

Running the Python 3 script discussed here should result in output similar to that shown in Figure 8-4.

```
Org ID: 669910444571366839 and Org Name: Devnet Mqzx2KNBledI
Network ID: L_669910444571380911, Name: branch office,Tags: []
Device Model: MX100    ,Serial: QBSA-BGUE-3A8T,MAC: 4c:c8:a1:02:00:1b, Firmware:Not running configured version
Device Model: MR52     ,Serial: QBSC-7GQT-R4C9,MAC: 4c:c8:a1:0c:01:44, Firmware:Not running configured version
Device Model: MV12W    ,Serial: QBSD-8G2Q-AS43,MAC: 4c:c8:a1:0f:01:36, Firmware:Not running configured version
```

Figure 8-4 *Output of the Python Script from Example 8-5*

Cisco Catalyst Center

Cisco Digital Network Architecture (DNA) is an open, extensible, software-driven architecture from Cisco that accelerates and simplifies enterprise network operations. Behind this architecture is the concept of intent-based networking, in which the network becomes an integral and differentiating part of the business. With Cisco DNA and the products behind it, network administrators define business intents that get mapped into infrastructure configurations by a central SDN controller. In the future, an intent-based network will dynamically adjust itself based on what it continuously learns from the traffic it transports as well as the business inputs it gets from the administrator.

Cisco Catalyst Center is the network management and command center for Cisco DNA. With Cisco Catalyst Center, you can provision and configure network devices in minutes, define a consistent policy throughout a network, get live and instantaneous statistics, and get granular networkwide views.

From a programmability perspective, Cisco Catalyst Center provides a set of REST APIs and SDKs through the Cisco Catalyst Center platform that are grouped in the following categories:

- Intent API
- Integration API
- Events and notifications

The Intent API is a northbound REST API that exposes specific capabilities of the Cisco Catalyst Center platform. The main purpose of the Intent API is to simplify the process of creating workflows that consolidate multiple network actions into one. An example is the SSID provisioning API, which is part of the Intent API. When you are configuring an SSID on a wireless network, several operations need to be completed, including creating a wireless interface, adding WLAN settings, and adding security settings. The SSID provisioning API combines all these operations and makes them available with one API call. This feature results in a drastic reduction in overall wireless SSID deployment time and also eliminates errors and ensures a consistent configuration policy. The Intent API provides automation capabilities and simplified workflows for QoS policy configuration, software image management and operating system updates for all devices in the network, overall client health status, and monitoring of application health. Application developers and automation engineers can take advantage of this single northbound integration layer to develop tools and applications on top of the network.

One of the main goals of the Cisco Catalyst Center platform is to simplify and streamline end-to-end IT processes. The Integration API was created for exactly this purpose. Through this API, the Cisco Catalyst Center platform publishes network data, events, and notifications to external systems and at the same time can consume information from these connected systems. Integrations with IT service management systems like ServiceNow, BMC Remedy, and other ticketing systems are supported. Automatic ticket creation and assignment based on network issues that are flagged by Cisco Catalyst Center are possible. Cisco Catalyst Center can even suggest remediation steps based on the machine learning algorithms that are part of the assurance capabilities. You can see how a typical support workflow can be improved with the Cisco Catalyst Center platform in the future. Cisco Catalyst Center detects a network issue, automatically creates a ticket, and assigns it to the right support personnel, along with a possible solution to the issue. The support team reviews the ticket and the suggested solution and can approve—either immediately or during a maintenance window—the remediation of the problem that Cisco Catalyst Center suggested. IP Address Management (IPAM) integrations are also supported by the Integration API. It is possible to seamlessly import IP address pools of information from IPAM systems such as Infoblox and BlueCat into Cisco Catalyst Center. Synchronization of IP pool/subpool information between Cisco Catalyst Center and IPAM systems is also supported. Through the Integration API that Cisco Catalyst Center provides, developers can integrate with any third-party IPAM solution.

Data as a service (DaaS) APIs that are part of the Integration API allow Cisco Catalyst Center to publish insights and data to external tools such as Tableau and similar reporting solutions. IT administrators have the option to build dashboards and extract business-relevant insights.

The Integration API is also used for cross-domain integrations with other Cisco products, such as Cisco Meraki, Cisco Secure Network Analytics, and Cisco ACI. The idea is to deliver a consistent intent-based infrastructure across the data center, WAN, and security solutions.

Cisco Catalyst Center also provides webhooks for events and notifications that are generated in the network or on the Cisco Catalyst Center appliance itself. You have the option of configuring a receiving URL to which the Cisco Catalyst Center platform can publish events. Based on these events, the listening application can take business actions. For instance, if some of the devices in a network are out of compliance, the events that the platform generates can be interpreted by a custom application, which might

trigger a software upgrade action in Cisco Catalyst Center. This completes the feedback loop in the sense that a notification generated by Cisco Catalyst Center is interpreted by a third-party custom application and acted upon by sending either an Intent API or Integration API call back to Cisco Catalyst Center to either remedy or modify the network, based on the desired business outcome. This mechanism of publishing events and notifications also saves on processing time and resources; before this capability existed, the custom application had to poll continuously to get the status of an event. Because the application subscribes through the webhook, polling can now be avoided entirely, and the custom application receives the status of the event right when it gets triggered.

Next, let's focus on the Cisco Catalyst Center Platform Intent API. As of this writing, the Intent API is organized into several distinct categories:

- **Know Your Network category:** This category contains API calls pertaining to sites, networks, devices, and clients:
  - With the Site Hierarchy Intent API, you can get information about, create, update, and delete sites as well as assign devices to a specific site. (Sites within Cisco Catalyst Center are logical groupings of network devices based on a geographic location or site.)
  - The Network Health Intent API retrieves data regarding network devices, their health, and how they are connected.
  - The Network Device Detail Intent API retrieves detailed information about devices. Different parameters can be passed to limit the scope of the information returned by the API, such as timestamp, MAC address, and UUID. Besides all the detailed information you can retrieve for all the devices in the network, you can also add, delete, update, or sync specified devices.
  - The Client Health Intent API returns overall client health information for both wired and wireless clients.
  - The Client Detail Intent API returns detailed information about a single client.
- **Site Management category:** This category helps provision enterprise networks with zero-touch deployments and manage the activation and distribution of software images in the network:
  - The Site Profile Intent API gives you the option to provision NFV and ENCS devices and retrieve the status of the provisioning activities.
  - The *Software Image Management (SWIM) API* enables you to completely manage the lifecycle of software images running within a network in an automated fashion. With this API, you can retrieve information about available software images, import images into Cisco Catalyst Center, distribute images to devices, and activate software images that have been deployed to devices.
  - The *Plug and Play (PnP) API* enables you to manage all PnP-related workflows. With this API, you can create, update, and delete PnP workflows and PnP server profiles, claim and unclaim devices, add and remove virtual accounts, and retrieve information about all PnP-related tasks.
- **Connectivity category:** This category contains APIs that provide mechanisms to configure and manage both non-fabric wireless and Cisco SDA wired fabric devices. For fabric devices, you can add and remove border devices to the fabric and get details about their status. For non-fabric wireless devices, you can create, update, and delete wireless SSIDs, profiles, and provisioning activities.
- **Operational Tools category:** This category includes APIs for the most commonly used tools in the Cisco Catalyst Center toolbelt:
  - The Command Runner API enables the retrieval of all valid keywords that Command Runner accepts and allows you to run read-only commands on devices to get their real-time configuration.
  - The Network Discovery API provides access to the discovery functionalities of Cisco Catalyst Center. You can use this API to create, update, delete, and manage network discoveries and the credentials needed for them. You can also retrieve network discoveries, network devices that were discovered as part of a specific network discovery task, and credentials associated with these discoveries.
  - The Template Programmer API can be used to manage configuration templates. You can create, view, edit, delete, version, add commands, check contents for errors, deploy, and check the status of template deployments.
  - The Path Trace API provides access to the Path Trace application in Cisco Catalyst Center. Path Trace can be used to troubleshoot and trace application paths throughout the network and provide statistics at each hop. The API gives you access to initiating, retrieving, and deleting path traces.
  - The File API enables you to retrieve files such as digital certificates, maps, and SWIM files from Cisco Catalyst Center.
  - The Task API provides information about the network actions that are being run asynchronously. Each of these background actions can take from seconds to minutes to complete, and each has a task associated with it. You can query the Task API about the completion status of these tasks, get the task tree, retrieve tasks by their IDs, and so on.
  - The Tag API gives you the option of creating, updating, and deleting tags as well as assigning tags to specific devices. Tags are very useful in Cisco Catalyst Center; they are used extensively to group devices by different criteria. You can then apply policies and provision and filter these groups of devices based on their tags.

The Cisco Catalyst Center platform APIs are rate limited to 100 API requests per minute.

So far in this section, we've covered some of the APIs offered by Cisco Catalyst Center. Next, we will start exploring the Intent API, using Cisco Catalyst Center version 2.3.7.9 for the rest of the chapter. Because API resources and endpoints exposed by the Cisco Catalyst Center platform might change in future versions of the software, it is always best to start exploring the API documentation for any Cisco product at https://developer.cisco.com/docs/dna-center/.

In Cisco Catalyst Center version 2.3.7.9, make sure the REST API is enabled. Log in to Catalyst Center with a super-admin role account, navigate to **Platform > Manage > Bundles**, and make sure the Catalyst Center REST API bundle is enabled. The status of the bundle should be active, as shown in Figure 8-5.

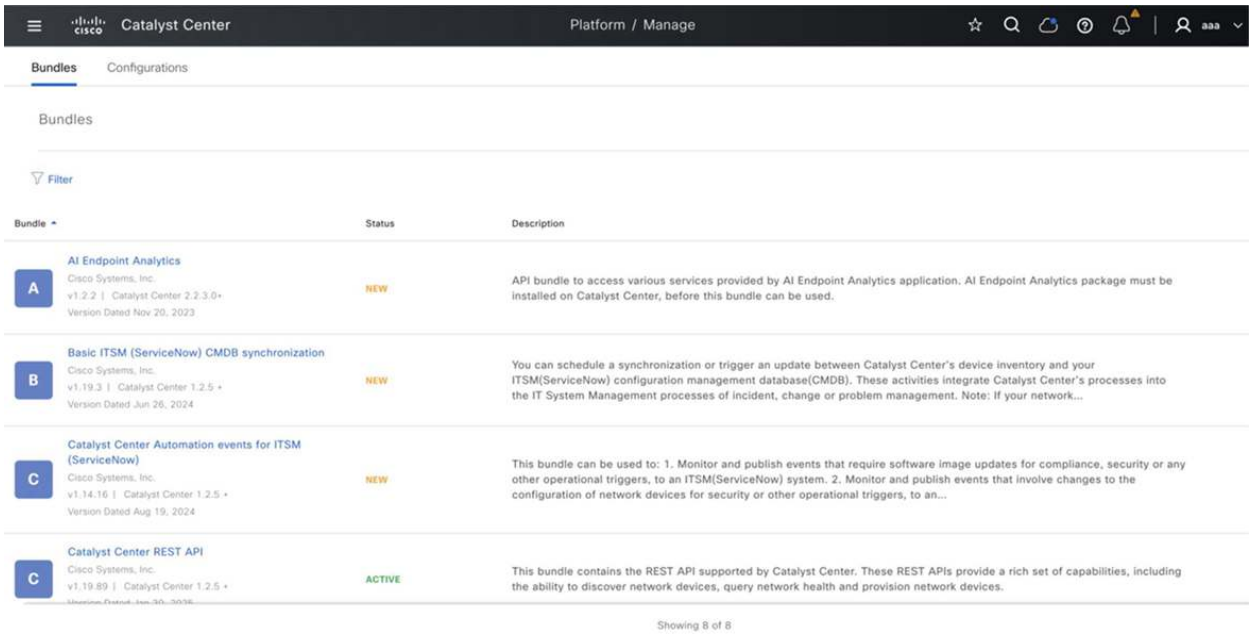


Figure 8-5 *Cisco Catalyst Center Platform Interface*

For this section, you can use the reservable DevNet Sandbox for Cisco Catalyst Center at https://devnetsandbox.cisco.com or your own Catalyst Center instance. If you reserved the DevNet sandbox, you could follow along or adjust the URL, username, and password for your own environment. After you get access to the sandbox, you can access Cisco Catalyst Center at https://10.10.20.85 with username **administrator** and password **Cisco1234!**. You need to get authorized to the API and get the token that you will use for all subsequent API calls. The Cisco Catalyst Center platform API authorization is based on basic auth. Basic auth, as you learned in Chapter 7, is an authorization type that requires a username and password to access an API endpoint. In the case of Cisco Catalyst Center, the username and password mentioned previously are base-64 encoded and then transmitted to the API service in the Authorization header. Many online services can do both encoding and decoding of base-64 data for you, or as a fun challenge, you can look up how to do it manually. The username **administrator** and the password **Cisco1234!** combined as **administrator:Cisco1234!** and base-64 encoded become **YWRtaW5pc3RyYXRvcjpDaXNjbzEyMzQh**. The only missing component is the resource that you need to send the authorization request to. You verify the documentation and see that the authorization resource is /dna/system/api/v1/auth/token and requires the API call to be POST. With this information, you can build the authorization API endpoint, which becomes https://10.10.20.85/dna/system/api/v1/auth/token.

Next, you will use curl, which is useful in testing REST APIs and web services. Armed with the authorization API endpoint, the Authorization header containing the base-64–encoded username and password, and the fact that the API call needs to be a POST call, you can now craft the authorization request in curl. The authorization API call with curl should look as follows:

```
curl -k -X POST /

 https://10.10.20.85/dna/system/api/v1/auth/token /

 -H 'Authorization: Basic YWRtaW5pc3RyYXRvcjpDaXNjbzEyMzQh'
```

The result should be JSON formatted with the key Token and a value containing the actual authorization token. It should look similar to the following:

```
{
 "Token": "eyJhbGciSnRl…[output omitted]…K2VQ",
 "message": ""
}
```

Next, let's explore how to obtain an authorization token using Postman. Make sure you select POST as the verb for the authorization API call and the endpoint https://10.10.20.85/dna/system/api/v1/auth/token. This is a POST call because you are creating new data in the system—in this case, a new authorization token. Under the Authorization tab, select **Basic Auth** as the type of authorization, and in the Username and Password fields, make sure you have the correct credentials (**administrator** and **Cisco1234!**). Since you have selected basic auth as the authorization type, Postman does the base-64 encoding for you automatically. All you need to do is click **Send**, and the authorization API call is sent to the specified URL. If the call is successfully completed, the status code should be 200 OK, and the body of the response should contain the JSON-formatted token key and the corresponding value. Figure 8-6 shows the Postman client interface with the information needed to successfully authenticate to the reservable Cisco DevNet Catalyst Center Sandbox.

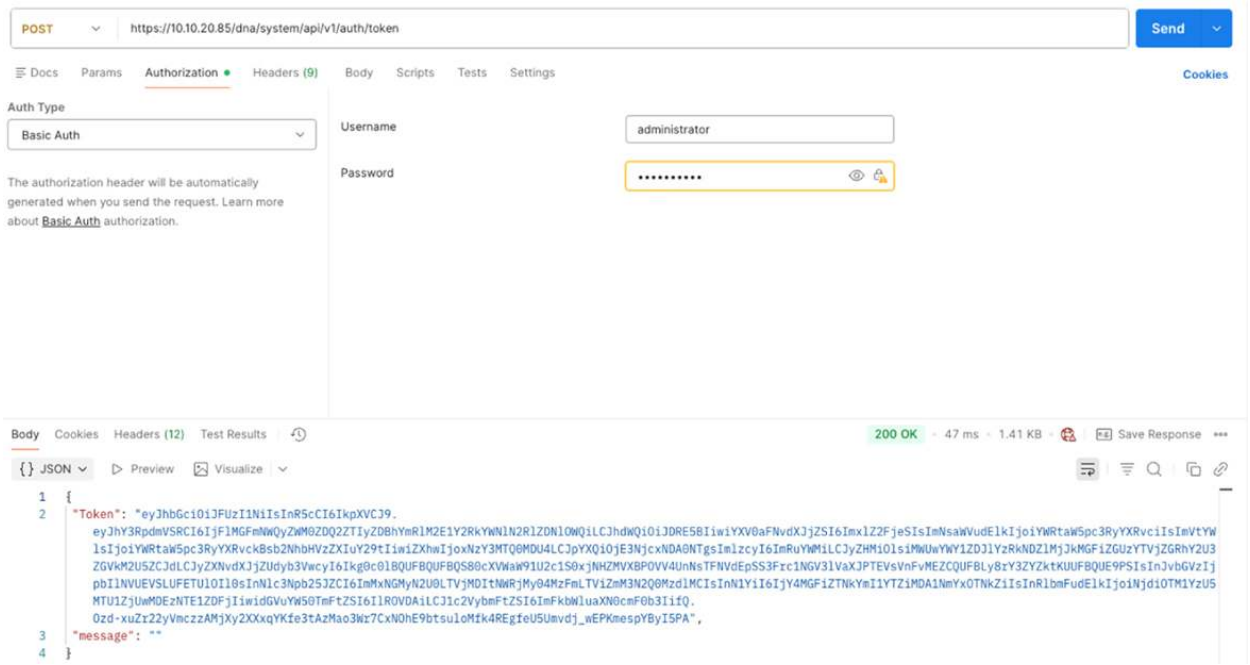


Figure 8-6 *Authenticating to Cisco Catalyst Center over the REST API*

The body of the response for the Postman request should look similar to the one you performed earlier with curl. As with the earlier curl example, this token will be used in all subsequent API calls performed in the rest of this chapter. The token will be passed in the API calls through a header that is called **X-Auth-Token**.

Let's now get a list of all the network devices that are managed by the instance of Cisco Catalyst Center that is running in the reservable DevNet Sandbox you've just authorized with. If you verify the Cisco Catalyst Center API documentation on https://developer.cisco.com/docs/dna-center/, you can see that the API resource that will return a complete list of all network devices managed by Cisco Catalyst Center is /dna/intent/api/v1/network-device. Figure 8-7 shows the online documentation for Cisco Catalyst Center version 2.3.7.9.

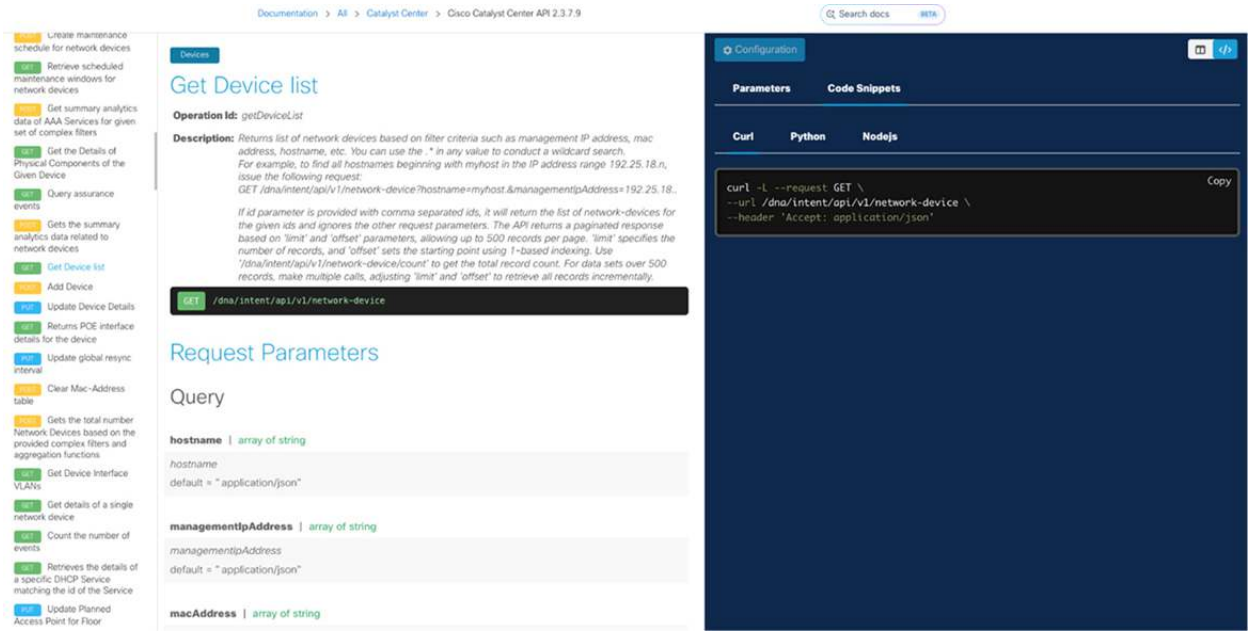


Figure 8-7 *Cisco Catalyst Center Platform API Documentation (https://developer.cisco.com)*

With all this information in mind, you can craft the curl request to obtain a list of all the network devices managed by the Cisco DevNet reservable Catalyst Center Sandbox. The complete URL is https://10.10.20.85/dna/intent/api/v1/network-device. You need to retrieve information through the API, so you need to do a GET request; don't forget the X-Auth-Token header containing the authorization token. The **curl** command should look as follows, and it should contain a valid token:

```
curl -k -X GET /

 https://10.10.20.85/dna/intent/api/v1/network-device /

  -H 'X-Auth-Token: eyJhbGciOiJFUzI1N…[output omitted]…K2VQ'
```

The response to this **curl** command should look as shown in Example 8-6.

Example 8-6 *List of Network Devices Managed by the Reservable Cisco Catalyst Center Sandbox Instance*


```
{
    "response": [
        {
            "memorySize": "NA",
            "family": "Switches and Hubs",
            "type": "Cisco Catalyst 9000 UADP 8 Port Virtual Switch",
            "lastUpdateTime": 1767140862889,
            "macAddress": "52:54:00:02:19:54",
            "softwareVersion": "17.12.1prd9",
            "deviceSupportLevel": "Supported",
            "softwareType": "IOS-XE",
            "serialNumber": "CML12345UAD",
            "collectionInterval": "Global Default",
            "dnsResolvedManagementAddress": "10.10.20.175",
            "lastManagedResyncReasons": "Periodic",
            "managementState": "Managed",
            "pendingSyncRequestsCount": "0",
            "reasonsForDeviceResync": "Periodic",
            "reasonsForPendingSyncRequests": "",
            … (output omitted)
        }
    ],
    "version": "1.0"
}
```

From this verbose response, you can extract some important information about the network devices managed by Cisco Catalyst Center. The data returned in the response is too verbose to fully include in the previous output, so just a snippet is included for your reference. As of this writing, in the complete response output, you can see that there are four Cisco Catalyst 9000 Series virtual switches in this network.

For each device, you can see extensive information such as the hostname, uptime, serial number, software version, management interface IP address, reachability status, hardware platform, and role in the network. You can see here the power of the Cisco Catalyst Center platform APIs. With one API call, you were able to get a complete status of all devices in the network. Without a central controller like Cisco Catalyst Center, it would have taken several hours to connect to each device individually and run a series of commands to obtain the same information that was returned with one API call in less than half a second. These APIs can save vast amounts of time and bring numerous possibilities in terms of infrastructure automation. The data returned by the API endpoints can be extremely large, so it might take a long time to process a request and return a complete response. As mentioned in Chapter 7, pagination is an API feature that allows for passing in parameters to limit the scope of the request. Depending on the Cisco Catalyst Center platform API request, different filter criteria can be considered, such as management IP address, MAC address, and hostname.

Now you will see how to obtain the same information you just got with curl but using Postman. The same API endpoint URL is used: https://10.10.20.85/dna/intent/api/v1/network-device. In this case, it is a GET request, and the X-Auth-Token header is specified under the Headers tab and populated with a valid token. If you click **Send** and there aren't any mistakes with the request, the status code should be 200 OK, and the body of the response should be similar to that obtained with the curl request. Figure 8-8 shows how the Postman interface should look in this case.

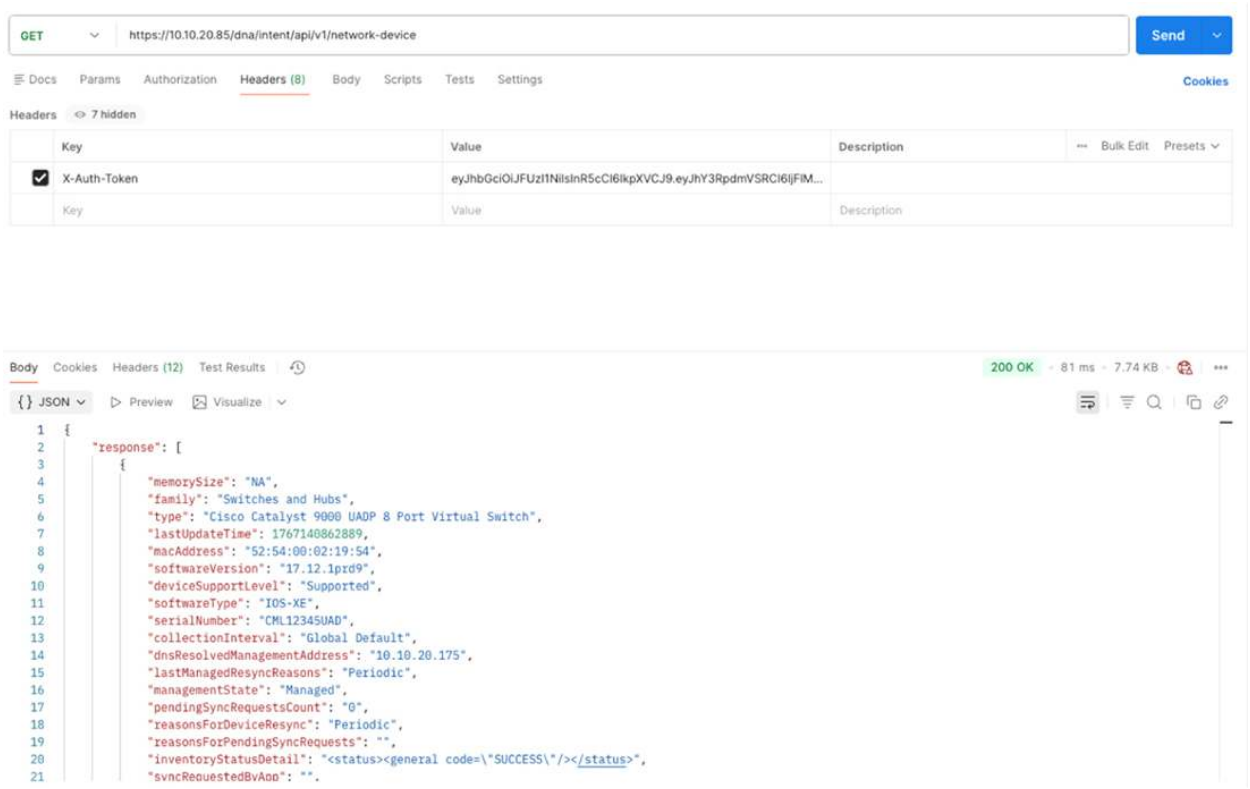


Figure 8-8 *Getting a List of Network Devices*

Now you can try to obtain data about the clients that are connected to the network managed by Cisco Catalyst Center. Much like network devices, network clients have associated health scores, provided through the Assurance feature to get a quick overview of client network health. This score is based on several factors, including onboarding time, association time, signal-to-noise ratio (SNR), and received signal strength indicator (RSSI) values for wireless clients, authentication time, connectivity and traffic patterns, and number of DNS requests and responses. In the API documentation, you can see that the resource providing the health status of all clients connected to the network is /dna/intent/api/v1/client-health. This API call offers the option to specify a parameter when performing the call. This parameter, called timestamp, represents the UNIX epoch time in milliseconds. UNIX epoch time is a system for describing a point in time since January 1, 1970, not counting leap seconds. It is used extensively in UNIX and many other operating systems. The timestamp provides the point in time for which the client health information should be returned in the API response. For example, if I retrieved the health status of all the clients connected to the network on Tuesday, December, 2025 5:00:00 PM PDT, the UNIX time, in milliseconds, would be 1767142747000. Keep in mind that based on the data retention policy set in Cisco Catalyst Center, client data might not be available for past distant timeframes.

With the information you now have, you can build the API endpoint to process the API call: https://10.10.20.85/dna/intent/api/v1/client-health. The timestamp parameter is not mandatory, so if no value is specified for it, the API call will return the client-health information at the current time. The authorization token also needs to be included in the call as a value in the X-Auth-Token header. The **curl** command should look as follows:

```
curl -k -X GET /

   https://10.10.20.85/dna/intent/api/v1/client-health/

   -H 'X-Auth-Token: eyJhbGciOiJFUzI1NiIs…[output omitted]…2VQ'
```

The API response should look similar to Example 8-7.

Example 8-7 *List of Clients and Their Status*

```
{
    "version": "1.0",
    "response": [
        {
            "siteId": "global",
            "scoreDetail": [
                {
                    "scoreCategory": {
                        "scoreCategory": "CLIENT_TYPE",
                        "value": "ALL"
                    },
                    "scoreValue": -1,
                    "clientCount": 0,
                    "clientUniqueCount": 0,
                    "maintenanceAffectedClientCount": null,
                    "randomMacCount": null,
                    "duidCount": null,
                    "starttime": 1767141927438,
                    "endtime": 1767142227438,
                    "connectedToUdnCount": 0,
                    "unconnectedToUdnCount": 0
                },... output omitted
}
```

From this response, you can see that there are no clients in this network. Another API endpoint that is useful to further investigate the health scores for individual clients is /dna/intent/api/v1/client-detail. This API call takes as an input parameter the MAC address of the client, which is a mandatory parameter, and the timestamp parameter, which is optional, and it returns extensive data about the status and health of that specific client at that specific time. If no value is specified for the timestamp, the detailed status of the client at the current time is returned.

Now you can try to perform the same API call but this time with Postman. The API endpoint stays the same: https://10.10.20.85/dna/intent/api/v1/client-health. In this case, you are trying to retrieve information from the API, so it will be a GET call, and the X-Auth-Token header contains a valid token value. Click **Send**, and if there aren't any errors with the API call, the body of the response should be similar to the one obtained previously with curl. The Postman window for this example should look as shown in Figure 8-9.

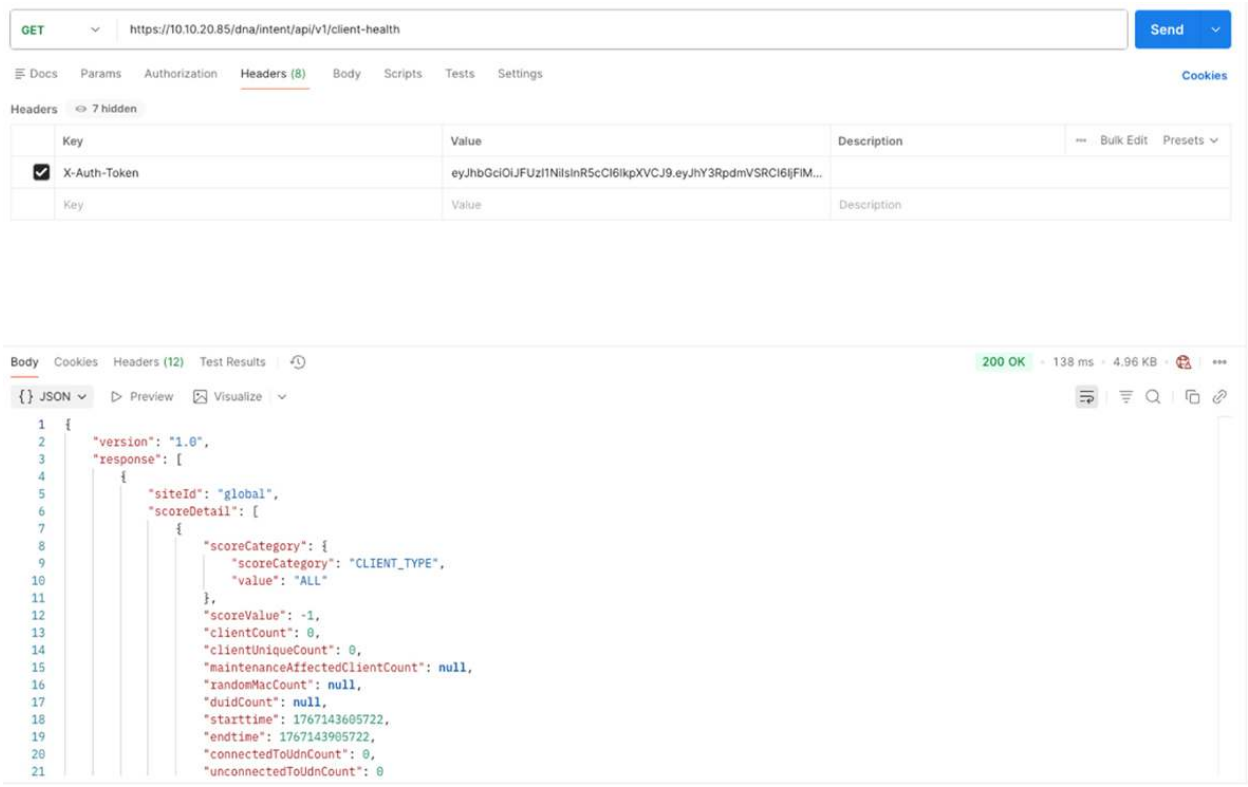


Figure 8-9 *Viewing Client Health in Cisco Catalyst Center*

So far in this section, we've explored the Cisco Catalyst Center platform API and seen how to authorize the API calls and obtain a token, how to get a list of all the devices in the network, and how to get health statistics for all the clients in the network using both curl and Postman. Next let's explore the Cisco Catalyst Center Python SDK. The SDK has been developed for Python 3 and maps all the Cisco Catalyst Center APIs into Python classes and methods to make it easier for developers to integrate with and expand the functionality of Cisco Catalyst Center. Installing the SDK is as simple as issuing the command **pip install catalystcentersdk** from the command prompt. At this writing, the current Cisco Catalyst Center SDK version is 3.1.3.0. The code in Example 8-8 was developed using this version of the SDK and Python 3.12.9. The code in Example 8-8 uses the SDK to first authorize to the API and then retrieve a list of all the network devices and client health statuses.

Example 8-8 *Python Script That Exemplifies the Use of the Cisco Catalyst Center Python SDK*

```
#! /usr/bin/env python
from catalystcentersdk import api

# Create a CatalystCenterAPI connection object;
# it uses Catalyst Center sandbox URL, username and password
CCenter = api.CatalystCenterAPI(username='administrator',
          password='Cisco1234!',
          base_url='https://10.10.20.85',
          verify=False)

# Find all devices
DEVICES = CCenter.devices.get_device_list()

# Print select information about the retrieved devices
print('{0:20s}{1:1}{2:50s}{3:1}{4:15s}'.format('Device Name', '|', \
 'Device Type', '|', 'Up Time'))
print('-'*95)
for DEVICE in DEVICES.response:
    print('{0:20s}{1:1}{2:50s}{3:1}{4:15s}'.format(DEVICE.hostname, \
        '|', DEVICE.type, '|', DEVICE.upTime))
print('-'*95)

# Get the health of all clients
CLIENTS = CCenter.clients.get_overall_client_health()

# Print select information about the retrieved client health statistics
print('{0:20s}{1:1}{2:50s}{3:1}{4:15s}'.format('Client Category', '|',\
 'Number of Clients', '|', 'Clients Score'))
print('-'*95)
for CLIENT in CLIENTS.response:
    for score in CLIENT.scoreDetail:
        print('{0:20s}{1:1}{2:<50d}{3:1}{4:<15d}'.format(
           score.scoreCategory.value, '|', score.clientCount, '|', \
           score.scoreValue))
print('-'*95)
```

First, this example imports the **api** class from **catalystcentersdk**. Next, it instantiates the **api** class and creates a connection to the reservable Cisco DevNet Sandbox Catalyst Center instance and stores the result of that connection in a variable called **CCenter**. If the connection was successfully established, the **CCenter** object has all the API endpoints mapped to methods that are available for consumption. **CCenter.devices.get_device_list()** provides a list of all the devices in the network and stores the result in the **DEVICES** dictionary. The same logic applies to the client health status. **CCenter.clients.get_overall_client_health()** returns a dictionary of health statuses for all the clients. When the data is extracted from the API and stored in the variables named **DEVICES** for all the network devices and **CLIENTS** for all the client health statuses, a rudimentary table is displayed in the console, with select information from both dictionaries. For the **DEVICES** variable, only the device name, device type, and device uptime are displayed, and for the **CLIENTS** variable, only the client health category, number of clients, and client score for each category are displayed.

The output of the Python script should look similar to that in Figure 8-10.

```
Device Name          |Device Type                                          |Up Time
-----------------------------------------------------------------------------------------------
sw1                  |Cisco Catalyst 9000 UADP 8 Port Virtual Switch       |1 day, 0:08:51.00
sw2                  |Cisco Catalyst 9000 UADP 8 Port Virtual Switch       |1 day, 0:08:41.00
sw3                  |Cisco Catalyst 9000 UADP 8 Port Virtual Switch       |1 day, 0:08:45.00
sw4                  |Cisco Catalyst 9000 UADP 8 Port Virtual Switch       |1 day, 0:08:35.00
-----------------------------------------------------------------------------------------------
Client Category      |Number of Clients                                    |Clients Score
-----------------------------------------------------------------------------------------------
ALL                  |0                                                    |-1
WIRED                |0                                                    |-1
WIRELESS             |0                                                    |-1
-----------------------------------------------------------------------------------------------
```

Figure 8-10 *Output of the Python Script from Example 8-8*

Cisco Catalyst SD-WAN

Cisco Software-Defined Wide Area Network (SD-WAN) is a cloud-first architecture for deploying WAN connectivity. Wide-area networks have been deployed for a long time, and many lessons and best practices have been learned throughout the years. Applying all these lessons to software-defined networking (SDN) resulted in the creation of Cisco SD-WAN. An important feature of SDN is the separation of the control plane from the data plane.

The control plane includes a set of protocols and features that a network device implements so that it can determine which network path to use to forward data traffic. Spanning tree protocols and routing protocols such as *Open Shortest Path First (OSPF), Enhanced Interior Gateway Routing Protocol (EIGRP),* and *Border Gateway Protocol (BGP)* are some of the protocols that make up the control plane in network devices. These protocols help build the switching or routing tables in network devices to enable them to determine how to forward network traffic.

The data plane includes the protocols and features that a network device implements to forward traffic to its destination as quickly as possible. *Cisco Express Forwarding (CEF)* is a proprietary switching mechanism that is part of the data plane. It was developed specifically to increase the speed with which data traffic is forwarded through network devices. You can read more about the control and data planes in Chapter 17, "Networking Components."

Historically, the control plane and data plane were part of the network device architecture, and they worked together to determine the path that the data traffic should take through the network and how to move this traffic as fast as possible from its source to its destination. As mentioned previously, *software-defined networking (SDN)* suggests a different approach.

SDN separates the functionality of the control plane and data plane in different devices, and several benefits result. First, the cost of the resulting network should be lower, because not all network devices have to implement expensive software and hardware features to accommodate both a control plane and data plane. The expensive intelligence from the control plane is constrained to a few devices that become the brains of the network, and the data plane is built with cheaper devices that implement only fast forwarding. Second, the convergence of this new network, which is the amount of time it takes for all devices to agree on a consistent view of the network, should be much lower than in the case of the non-SDN architectures of the past. In networks of similar sizes, the ones built with network devices that implement both the control plane and the data plane in their architecture take much longer to exchange all the information needed to forward data traffic than do the networks that implement separate control and data plane functionality in their architecture. Depending on the size of a network, this could mean waiting for thousands of devices to exchange information through their control plane protocols and settle on a certain view of the network or wait for tens of SDN controllers to accomplish the same task; the convergence time improvements are massive.

Cisco currently has two SD-WAN offerings. The first one is called Cisco Catalyst SD-WAN, and the second one, based on the Meraki acquisition, is called Meraki SD-WAN. We already covered Cisco Meraki earlier in this chapter; this section covers Cisco Catalyst SD-WAN.

You've already seen some of the advantages that SDN brings to WAN connectivity. Based on this new architecture and paradigm, the Cisco Catalyst SD-WAN offering contains several products that perform different functions:

- **SD-WAN Manager:** Cisco SD-WAN Manager is a centralized network management system that provides a GUI and REST API interface to the SD-WAN fabric. You can easily manage, monitor, and configure all Cisco SD-WAN components through this single pane of glass.
- **SD-WAN Controller:** Cisco SD-WAN Controller is the brains of the centralized control plane for the overlay SD-WAN network. It maintains a centralized routing table and centralized routing policy that it propagates to all the network Edge devices through permanent DTLS tunnels.
- **SD-WAN Validator:** Cisco SD-WAN Validator is the orchestrator of the fabric. It authenticates the SD-WAN Controllers and the Edge devices and coordinates connectivity between them. The SD-WAN Validator is the only component in the SD-WAN fabric that needs public IP reachability to ensure that all devices can connect to it.
- **WAN Edge:** Cisco WAN Edge routers, as the name implies, are Edge devices that are located at the perimeter of the fabric, such as in remote offices, data centers, branches, and campuses. They represent the data plane and bring the whole fabric together and route traffic to and from their site across the overlay network.

All the components of the Cisco Catalyst SD-WAN fabric run as virtual appliances, and the WAN Edges are also available as hardware routers.

Separating the WAN fabric this way makes it more scalable, faster to converge, and cheaper to deploy and maintain. On top of a transport-independent underlay that supports all types of WAN circuits (MPLS, DSL, broadband, 5G, and so on), an overlay network is being built that runs *Overlay Management Protocol (OMP).* Much like BGP, OMP propagates throughout the network all the routing information needed for all the components of the fabric to be able to forward data according to the routing policies configured in SD-WAN Manager.

Cisco SD-WAN Manager provides a REST API interface that exposes the functionality of the Cisco SD-WAN software and hardware features. The API resources that are available through the REST interface are grouped in the following collections:

- **Administration:** For management of users, groups, and local SD-WAN Manager instances
- **Certificate Management:** For management of TLS certificates and security keys
- **Configuration:** For creation of feature and device configuration templates and creation and configuration of SD-WAN Manager clusters
- **Device Inventory:** For collecting device inventory information, including system status
- **Monitoring:** For getting access to status, statistics, and related operational information about all the devices in the network every 10 minutes from all devices
- **Real-Time Monitoring:** For gathering real-time monitoring statistics and traffic information approximately once per second
- **Troubleshooting Tools:** For API calls used in troubleshooting, such as to determine the effects of applying a traffic policy, updating software, or retrieving software version information

Cisco SD-WAN Manager exposes a self-documenting web interface for the REST API, based on the OpenAPI specification. This web interface is enabled by default and can be accessed at https://SD-WAN_Manager_IP_or_hostname:port/apidocs. SD-WAN_Manager_IP_or_hostname is the IP address or hostname of the Cisco SD-WAN Manager server, and the port is 443 by default. The rest of this chapter uses the reservable Cisco DevNet SD-WAN Sandbox, available at https://devnetsandbox.cisco.com. After you reserve the sandbox, you can reach the Cisco SD-WAN Manager GUI at https://10.10.20.90 with username **admin** and password **C1sco12345**. At this writing, this sandbox is running Cisco Catalyst SD-WAN version 20.18.1 for all components of the fabric. Because this sandbox will be updated in time, or you might be using a different version, you should check https://developer.cisco.com for the latest information on all Cisco REST API documentation and changes. After you specify the credentials, the self-documenting web interface of https://10.10.20.90/apidocs looks as shown in Figure 8-11.

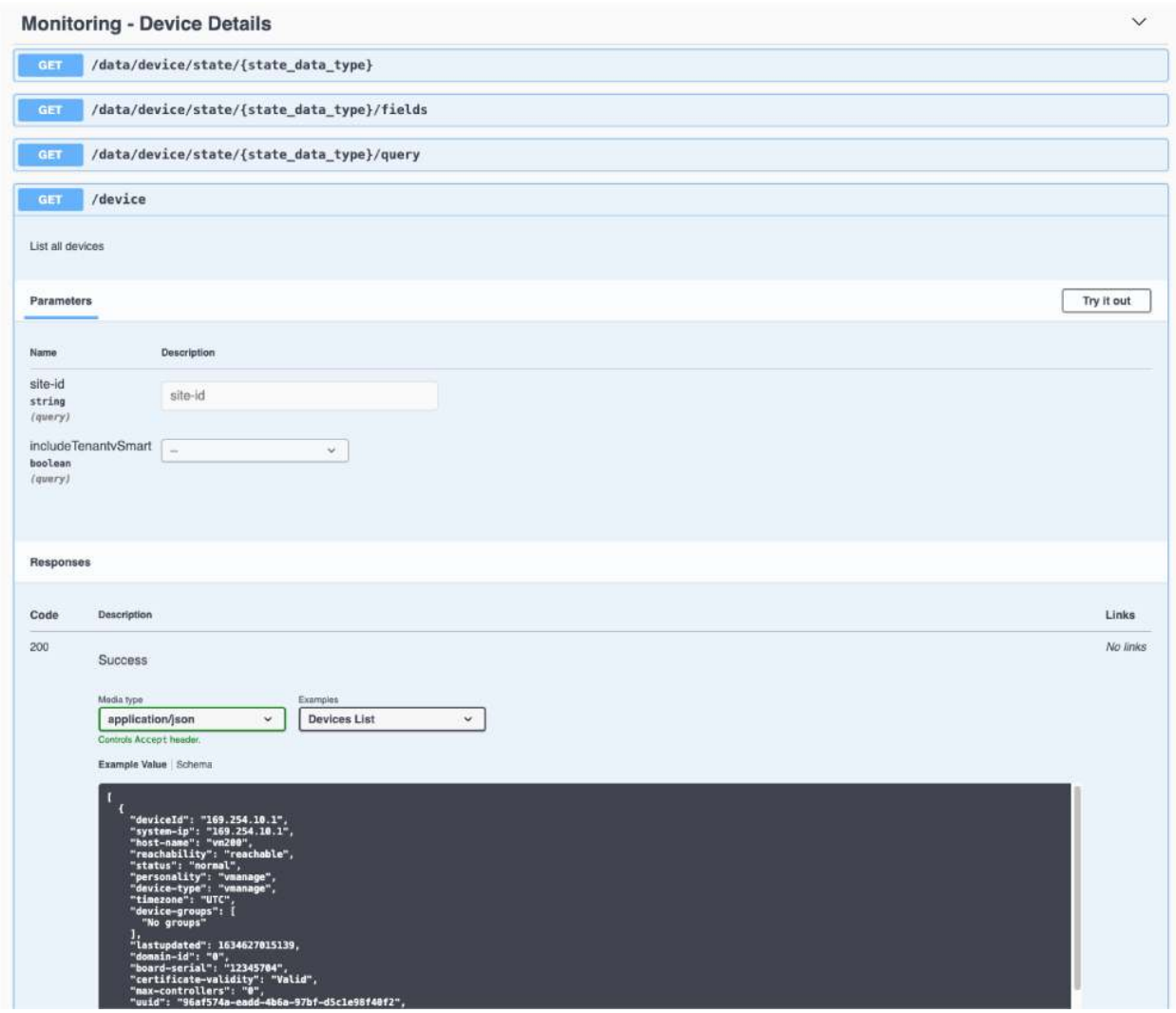


Figure 8-11 *Cisco Catalyst SD-WAN OpenAPI Specification-Based Interface*

This web interface displays all the REST API resources available, the methods associated with each one of them, and the model schema of the responses. The option of trying out each API call and exploring the returned data is also available.

Let's explore the Cisco SD-WAN Manager REST API next. You can find the API documentation at https://developer.cisco.com/docs/sdwan/. At this link, you can find all the information needed on how to interact with the REST

API, all the resources available, and extensive explanations.

Starting from release 20.18, Cisco Catalyst SD-WAN Manager supports JSON web token (JWT) authentication for REST API access control. Cisco Catalyst SD-WAN Manager continues to support session-based authentication using JSESSIONIDs to ensure backward compatibility.

JWT-based authentication starts by logging in with a username and password to receive a JWT token. The authentication API call is a POST to the /jwt/login endpoint with content type application/json in the request header. The mandatory username and password attributes are passed as payload along with an optional duration attribute, expressed in seconds, for the access token in the POST request body. The default expiration duration for the access token is 1800 seconds (30 minutes). Here is an example using the DevNet sandbox reserved earlier:

```
curl -k -X POST /

 "https://10.10.20.90/jwt/login" /

 -H "Content-Type: application/json" /

 -d "{"username":"admin", "password":"C1sco12345"}"
```

When user authentication is successful, the response body contains a JSON object similar to that shown in Example 8-9.

Example 8-9 *Successful Authentication to the Cisco Catalyst SD-WAN Manager*

```
{
  "token": "eyJ0eXAiOiJqd3QiLCJhbGciOiJSUzI1NiJ9…[output ommited]…WShQ",
  "refresh": "eyJ0eXAiOiJqd3QiLCJhbGciOiJSUzI1NiJ9…[output omitted]…h9A",
  "sub": "admin",
  "iss": "8d104923-c5ff-4b6c-87e1-9ef229f39a17",
  "aud": "vmanage",
  "userGroup": "[netadmin]",
  "tenant": "cml-sdwan-lab-tool",
  "duration": 1800,
  "exp": 1768870371,
  "csrf":
"E0718180EFE5F867B37EFA087714F54764DFEC9BB302B933536B184553E39907988666D14E8F
4F791AA8A84D3816AFAB28AC",
  "isAPIKey": false,
  "tenantId": "default"
}
```

Within the JSON object, you can retrieve the access token through the token attribute within the response body. The csrf attribute contains a cross-site request forgery (XSRF) prevention token, which is necessary for most POST operations. Apart from the access token, Cisco Catalyst SD-WAN Manager also returns a refresh token. The refresh token is available in the refresh attribute of the response. The system sets the refresh token expiration duration to one day multiplied by the duration of the access token. You can use the refresh token to generate a new access token without presenting the userid and password credentials through a POST call to the /jwt/refresh API endpoint.

Figure 8-12 shows the same API call in Postman.

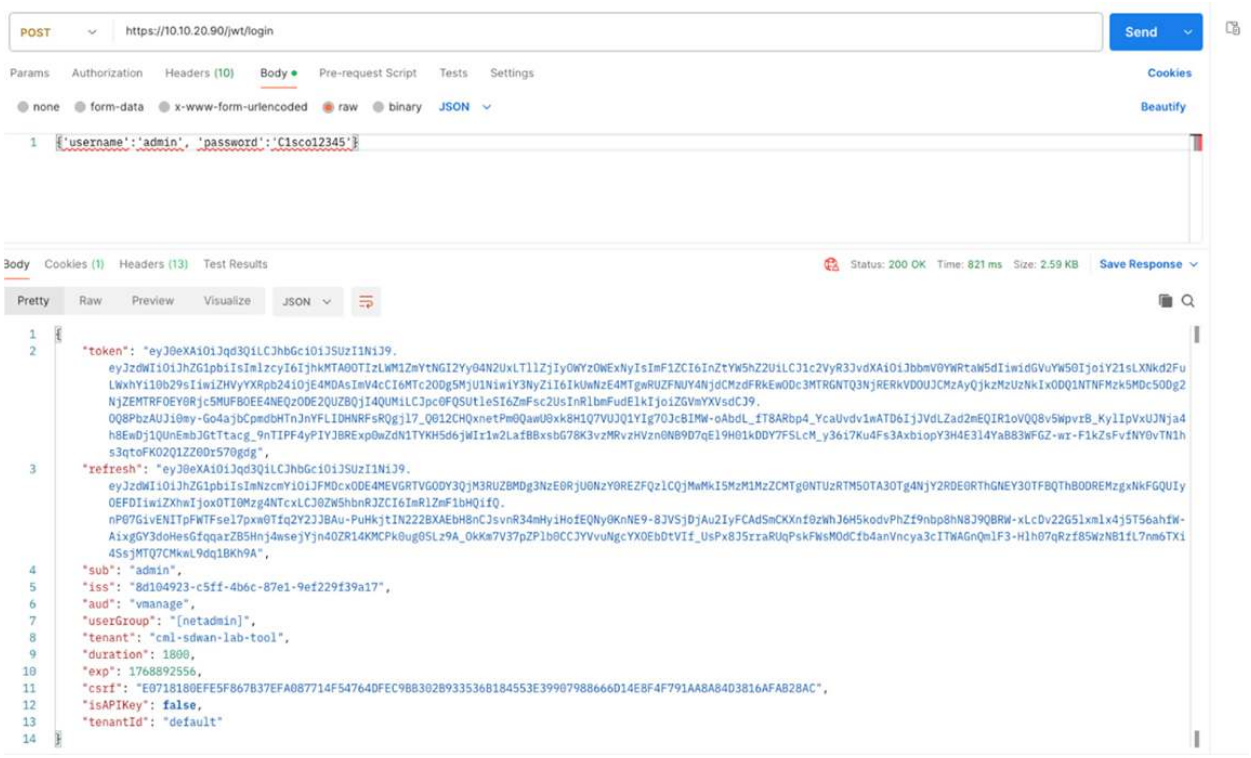


Figure 8-12 *Cisco Catalyst SD-WAN REST API Authorization Call*

The status code of the response should be 200 OK, and the body of the response should be similar to the one you received when performing the authorization call using curl.

For calls to API endpoints, you must provide the access token within the Authorization header of the HTTP request in the form of a bearer token. For POST requests, you must also provide the XSRF token in a header called X-XSRF-TOKEN. To get a list of all the devices in the SD-WAN fabric managed by the Catalyst SD-WAN Manager from the DevNet sandbox, you would use the **curl** command like this:

```
curl -k -X GET 'https://10.10.20.90/dataservice/device' /

-H 'Content-Type: application/json' /

-H 'Authorization: Bearer eyJ0eXAiOiJqd…[output ommited]…TpQ'
```

The response should look similar to the one in Example 8-10.

Example 8-10 *List of Devices That Are Part of the Cisco Catalyst SD-WAN Fabric*

```
{
   ... omitted output
   "data" : [
    {
      "deviceId": "100.0.0.1",
      "system-ip": "100.0.0.1",
      "host-name": "Manager01",
      "reachability": "reachable",
      "status": "normal",
      "personality": "vmanage",
      "device-type": "vmanage",
      "device-groups": [
        "No groups"
      ],
      "lastupdated": 1768871910300,
      "domain-id": "0",
      "board-serial": "2940B5BB489B4B139700858495F1F3A7",
      "certificate-validity": "Valid",
      "max-controllers": "0",
      "uuid": "8d104923-c5ff-4b6c-87e1-9ef229f39a17",
      "controlConnections": "6",
      "device-model": "vmanage",
      "version": "20.18.1",
      "connectedVManages": [
        "100.0.0.1"
      ],
      "site-id": "100",
      "latitude": "37.666684",
      "longitude": "-122.777023",
      "isDeviceGeoData": false,
      "platform": "x86_64",
      "uptime-date": 1768568711295,
      "statusOrder": 4,
      "device-os": "next",
      "validity": "valid",
      "state": "green",
      "state_description": "All daemons up",
      "model_sku": "None",
      "local-system-ip": "100.0.0.1",
      "total_cpu_count": "10",
      "testbed_mode": false,
      "layoutLevel": 1,
      "site-name": "SITE_100"
    },
    ... omitted output
   ]
}
```


The output of this GET API call is too verbose to be fully displayed. We encourage you to explore this API call and observe the full response on your own computer.

The body of the response is in JSON format and contains information about all the devices in the SD-WAN fabric. As of this writing, this specific fabric contains the following:

- One Cisco SD-WAN Manager server
- One Cisco SD-WAN Controller server
- One Cisco SD-WAN Validator server
- Four Cisco WAN Edge routers

For each device, the response includes status, geographic coordinates, role, device ID, uptime, site ID, SSL certificate status, and more.

You can build the same request in Postman and send it to SD-WAN Manager as shown in Figure 8-13. The body of the response is similar to the one received from the **curl** command.

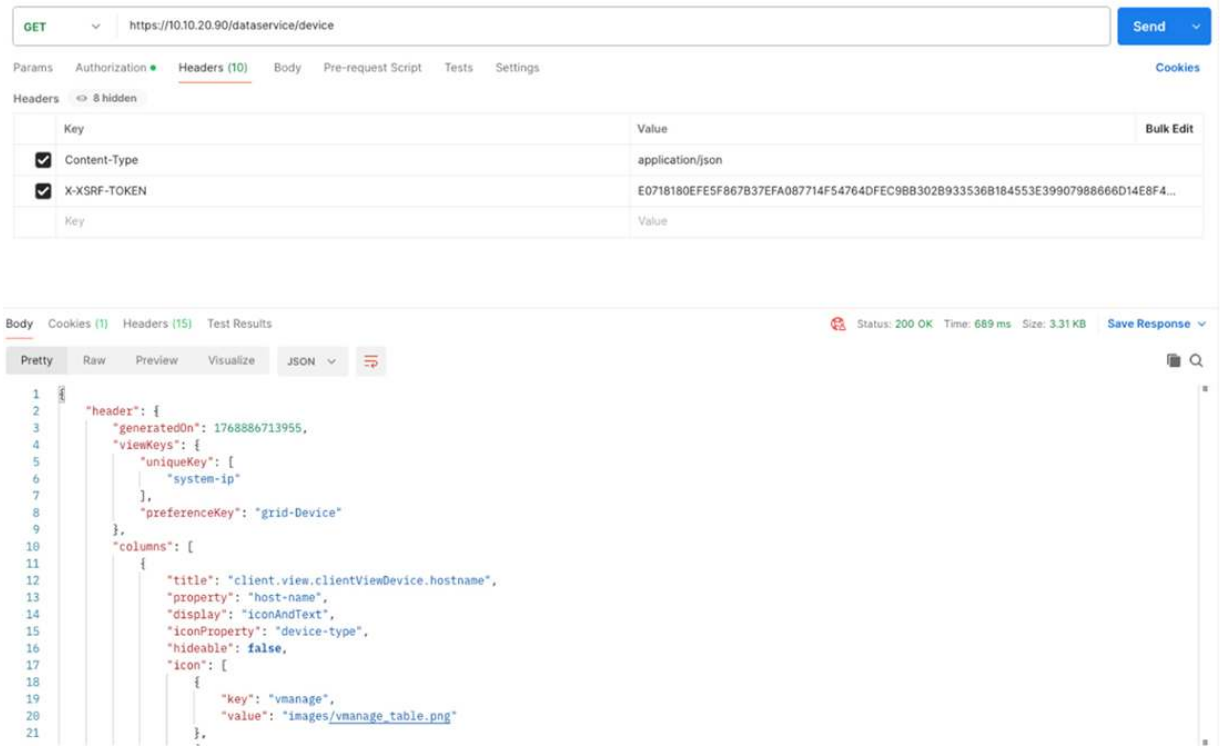


Figure 8-13 *Getting a List of All the Devices in the Cisco Catalyst SD-WAN Fabric*

While exploring the Cisco Catalyst SD-WAN REST API, let's get a list of all the device templates that are configured on the Cisco DevNet Manager server. According to the API documentation, the resource that will return this information is /dataservice/template/device. You pass in the JSESSIONID value in the cookie header and build the following **curl** command:

```
curl -k -X GET /

  https://10.10.20.90/dataservice/template/device /

  -H 'Content-Type: application/json' /

  -H 'Authorization: Bearer eyJ0eXAiOiJqd…[output ommited]…TpQ'
```

The response from the Manager server at https://10.10.20.90 should look as shown in Example 8-11.

Example 8-11 *List of Device Configuration Templates*

```
{
   "data" : [
      {
            "deviceType": "vedge-ISR-4331",
            "lastUpdatedBy": "system",
            "resourceGroup": "global",
            "deviceRole": "sdwan-edge",
            "templateClass": "cedge",
            "configType": "template",
            "templateId": "f240da35-56ba-444f-84c5-e39f910708fc",
            "factoryDefault": true,
            "templateName": "Factory_Default_ISR_4331_V01",
            "devicesAttached": 0,
            "templateDescription": "Factory Default template for ISR4331 device",
            "lastUpdatedOn": 1768569272903,
            "templateAttached": 14,
            "draftMode": "Disabled"
      }
   ],
... output omitted
}
```


The response contains details about the device templates available on this Manager server. The same information is returned by Manager when using Postman to get the list of all device templates. Figure 8-14 shows the Postman client interface with the headers needed to retrieve a list of all the device configuration templates available on a specific Manager instance.

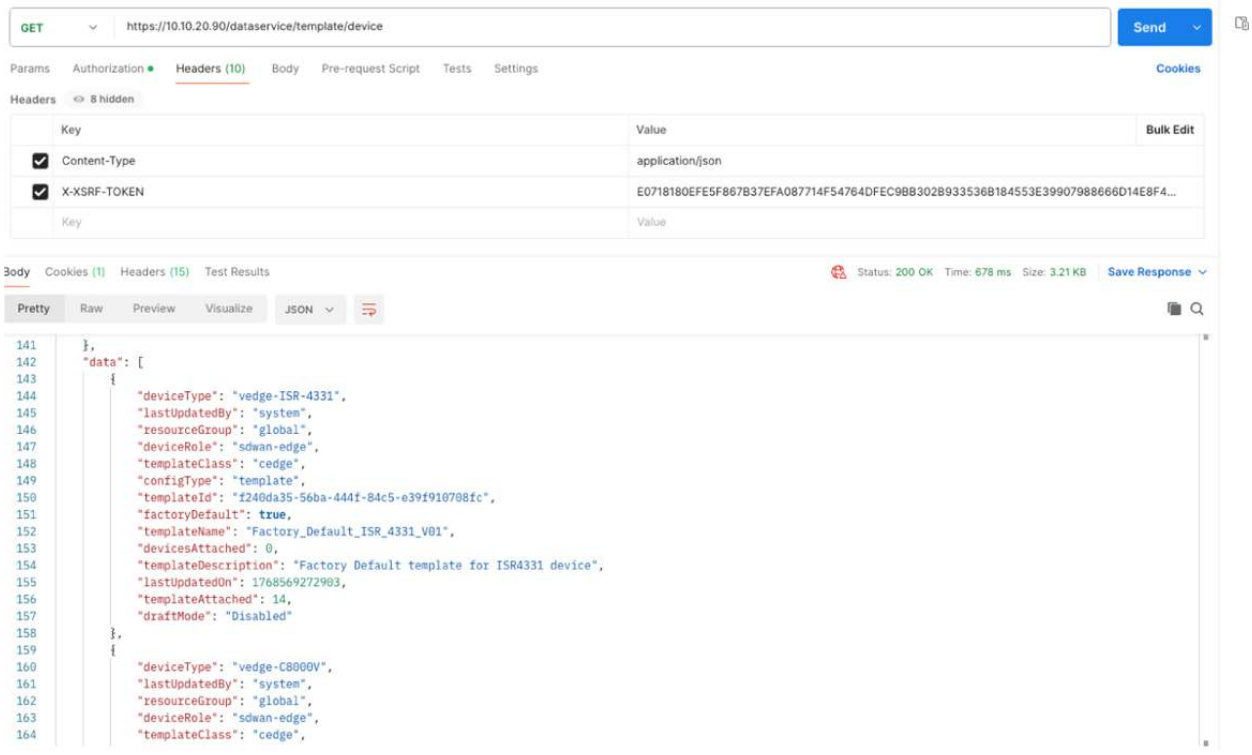


Figure 8-14 *Getting Device Configuration Templates*

Cisco Catalyst SD-WAN Manager still supports the traditional REST API access control that is based on sessions (JSESSIONIDs). With this authentication method, Cisco Catalyst SD-WAN Manager creates a session for each user after successful login. The initial connection is established through an authorization request to https://10.10.20.90/j_security_check. The information sent over this POST call is URL form encoded and contains the username and password mentioned previously. The curl request for this authorization request should look as follows:

```
curl -k -c - -X POST  /

 https://10.10.20.90/j_security_check /

 -H 'Content-Type: application/x-www-form-urlencoded' /

 -d 'j_username=admin&j_password=C1sco12345'
```

The **-c -** option passed to the curl request specifies that the returned authorization cookie should be printed to the console. The **-k** option bypasses TLS certificate verification, because the certificate for this sandbox is self-signed. The output of the command should look as follows:

```
# Netscape HTTP Cookie File

# https://curl.haxx.se/docs/http-cookies.html

# This file was generated by libcurl! Edit at your own risk.


#HttpOnly_10.10.20.90    FALSE. /        TRUE    0        JSESSIONID

         t8Y4Z2RrO5xf73HR-blA9weOq2S9ZxfqmNBlFeUJ.0ea9f09c2925a52a6ad90e75

   da496c52ae096b731fa80cc75aec4562887915c8
```

The long string after JSESSIONID is the value of the authorization cookie that will be needed in all subsequent API calls.

To see an example, you can try to get a list of all the devices that are part of this Cisco SD-WAN fabric. According to the documentation, the resource that will return this information is /dataservice/device. It will have to be a GET call, and the JSESSIONID cookie needs to be passed as a header in the request. The **curl** command to get a list of all the devices in the fabric should look like as follows:

```
curl -X GET -k /

 https://10.10.20.90/dataservice/device /

 -H 'Cookie:

  JSESSIONID=v9QcTVL_ZBdIQZRsI2V95vBi7Bz47IMxRY3XAYA6.4854266f-a8ad-4068-9

  651-d4e834384f51'
```

Next, let's use Python to build a script that will go through the same steps: Log in to Manager, get a list of all the devices in the SD-WAN fabric, and get a list of all device templates available. No SDK will be used in this case; this will help you see the difference between this code and the Python code you used earlier in this chapter. Since no SDK will be used, all the API resources, payloads, and handling of data will have to be managed individually.

The Python **requests** library will be used extensively in this sample code. You should be familiar with this library from Chapter 7. Example 8-12 shows a possible version of the Python 3 script that accomplishes these tasks. The script was developed using Python 3.13.2 and version 2.32.3 of the requests library. The **json** library that comes with Python 3.13.2 was also used to deserialize and load the data returned from the REST API into a Python object; in this case, that object is a list. In this code, first, the **import** keyword makes the two libraries **requests** and **json** available for use within the script. Since the connection in the script is made to an instance of Manager that is in a sandbox environment and that uses a self-signed TLS certificate, the third and fourth lines of the script disable the warning messages that are generated by the **requests** library when connecting to REST API endpoints that are secured with self-signed TLS certificates. Next, the script specifies the Manager hostname and the username and password for this instance; this example uses the same Manager server used earlier in this chapter. The code then specifies the base URL for the Manager REST API endpoint: https://10.10.20.90. The code shows the authentication resource (j_security_check) and the login credentials, and then the login URL is built as a combination of the base URL and the authentication API resource. In the next line, a new **request** session instance is created and stored in the **SESS** variable.

Example 8-12 *Python Script Showcasing How to Interact with the Cisco Catalyst SD-WAN REST API*

```
#! /usr/bin/env python
import json
import requests
from requests.packages.urllib3.exceptions import InsecureRequestWarning
requests.packages.urllib3.disable_warnings(InsecureRequestWarning)

# Specify Cisco Manager IP, username and password
MANAGER_IP = "10.10.20.90"
USERNAME = "admin"
PASSWORD = "C1sco12345"

BASE_URL_STR = "https://{}/".format(MANAGER_IP)

# Login API resource and login credentials
LOGIN_ACTION = "j_security_check"
LOGIN_DATA = {"j_username" : USERNAME, "j_password" : PASSWORD}


# URL for posting login data
LOGIN_URL = BASE_URL_STR + LOGIN_ACTION

# Establish a new session and connect to Cisco Manager
SESS = requests.session()
LOGIN_RESPONSE = SESS.post(url=LOGIN_URL, data=LOGIN_DATA, verify=False)

# Get list of devices that are part of the fabric and display them
DEVICE_RESOURCE = "dataservice/device"
# URL for device API resource
DEVICE_URL = BASE_URL_STR + DEVICE_RESOURCE

DEVICE_RESPONSE = SESS.get(DEVICE_URL, verify=False)
DEVICE_ITEMS = json.loads(DEVICE_RESPONSE.content)["data"]

print("{0:52s}{1:1}{2:20s}{3:1}{4:40s}{5:1}{6:16s}{7:1}{8:7s}"\
    .format('Host-Name', '|', 'Device Model', '|', 'Device ID', \
        '|', 'System IP', '|', 'Site ID'))
print("-"*150)

for ITEM in DEVICE_ITEMS:
     print("{0:52s}{1:1}{2:20s}{3:1}{4:40s}{5:1}{6:16s}{7:1}{8:7s}"\
         .format(ITEM['host-name'], '|', ITEM['device-model'], '|', \
               ITEM['uuid'], '|', ITEM['system-ip'], '|', ITEM['site-id']))
print("-"*150)
# Get list of device templates and display them
TEMPLATE_RESOURCE = "dataservice/template/device"
# URL for device template API resource
TEMPLATE_URL = BASE_URL_STR + TEMPLATE_RESOURCE

TEMPLATE_RESPONSE = SESS.get(TEMPLATE_URL, verify=False)
TEMPLATE_ITEMS = json.loads(TEMPLATE_RESPONSE.content)["data"]

print("{0:52s}{1:1}{2:20s}{3:1}{4:40s}{5:1}{6:16s}{7:1}{8:7s}"\
     .format('Template Name', '|', 'Device Model', '|', 'Template ID', \
         '|', 'Attached devices', '|', 'Template Version'))
print("-"*150)

for ITEM in TEMPLATE_ITEMS:
     print("{0:52s}{1:1}{2:20s}{3:1}{4:40s}{5:1}{6:<16d}{7:1}{8:<7d}"\
           .format(ITEM['templateName'], '|', ITEM['deviceType'], '|', \
               ITEM['templateId'], '|', ITEM['devicesAttached'], '|', \
```

```
                    ITEM['templateAttached']))
print("-"*150)
```

Using this new session, a POST request is sent to the login URL, containing the username and password as payload and disabling TLS certificate verification by specifying verify=False. At this point, a session is established to the DevNet Sandbox Manager instance. This session can be used to interact with the Manager REST API by getting, creating, modifying, and deleting data.

The code specifies the API resource that will return a list of all the devices in the SD-WAN fabric: dataservice/device. The complete URL to retrieve the devices in the fabric is built on the next line by combining the base URL with the new resource. The **DEVICE_URL** variable will look like https://10.10.20.90/dataservice/device. Next, the same session that was established earlier is used to perform a GET request to the device_url resource. The result of this request is stored in the variable aptly named **DEVICE_RESPONSE**, which contains the same JSON-formatted data that was obtained in the previous curl and Postman requests, with extensive information about all the devices that are part of the SD-WAN fabric. From that JSON data, only the list of devices that are values of the **data** key are extracted and stored in the **DEVICE_ITEMS** variable.

Next, the header of a rudimentary table is created. This header contains the fields Host-Name, Device Model, Device ID, System IP, and Site ID. From the extensive list of information contained in the **DEVICE_ITEMS** variable, only these five fields will be extracted and displayed to the console for each device in the fabric. The code next prints a series of delimiting dashes to the console to increase the readability of the rudimentary table. The next line of code has a **for** loop that is used to iterate over each element of the **DEVICE_ITEMS** list and extract the hostname, device model, device ID, system IP address, and site ID for each device in the fabric and then display that information to the console. The code then prints a series of dashes for readability purposes. Next, the same logic is applied to GET data from the API, but this time the data is gathered about all the device templates that are configured on this instance of Manager. The URL is built by concatenating the base URL with the device template resource, dataservice/template/device. The same session is reused once more to obtain the data from the REST API. In the case of the device templates, only the template name, the type of device the template is intended for, the template ID, the number of attached devices to each template, and the template version are extracted and displayed to the console.

If you run this script in a Python 3.13.2 virtual environment with the **requests** library version 2.32.3 installed, you get output similar to that shown in Figure 8-15.

```
Host-Name                                                       |Device Model       |Device ID                                |System IP       |Site ID
------------------------------------------------------------------------------------------------------------------------------------------------------
Manager01                                                       |vmanage            |8d104923-c5ff-4b6c-87e1-9ef229f39a17     |100.0.0.1       |100
Controller01                                                    |vsmart             |0b16ec65-504c-48bc-a258-a960bf1dc882     |100.0.0.101     |100
Validator01                                                     |vedge-cloud        |53dfce38-30fe-434d-82b3-8e66fe65c348     |100.0.0.201     |100
Edge1                                                           |vedge-C8000V       |C8K-3D1A8960-6E76-532C-DA93-50626FC5797E|10.0.0.1        |1
Edge2                                                           |vedge-C8000V       |C8K-D4CE7174-5261-7E6F-91EA-4926BCF4C2DD|10.0.0.2        |2
Edge3                                                           |vedge-C8000V       |C8K-27148AFE-7160-69D3-ACC5-F77560530D63|10.0.0.3        |3
Edge4                                                           |vedge-C8000V       |C8K-3188FA44-8A8A-8392-0E4B-0DA5A445C10B|10.0.0.4        |4
------------------------------------------------------------------------------------------------------------------------------------------------------
Template Name                                                   |Device Model       |Template ID                              |Attached devices|Template Version
------------------------------------------------------------------------------------------------------------------------------------------------------
Factory_Default_ISR_4331_V01                                    |vedge-ISR-4331     |f240da35-56ba-444f-84c5-e39f910708fc     |0               |14
Default_SDBranch_C8000V_Template_V01                            |vedge-C8000V       |3ec70b2d-096f-4ac4-9ba0-dcefcc952054     |0               |16
Default_AWS_TGW_C8000V_Template_V01                             |vedge-C8000V       |b0db2649-5285-499c-b2da-e30ea15ac7fe     |0               |11
Factory_Default_C1111_8PLTELA_V01                               |vedge-C1111-8PLTELA|4cf17488-592d-468b-964e-2111fd03adf4     |0               |15
Default_SDBranch_vEdge_cloud_Template_V01                       |vedge-cloud        |20834a53-04e8-46c7-843d-cd2b5347af3f     |0               |14
Factory_Default_DC1_HUB                                         |vedge-C8000V       |01cb4a76-05a5-4378-b155-e2d341dd0ef4     |0               |19
Default_AWS_TGW_CSR1000V_Template_V01                           |vedge-CSR-1000v    |48ff427e-01d3-48e5-b1f4-6b1e7b9c5947     |0               |11
Default_GCP_C8000V_Template_V01                                 |vedge-C8000V       |6ecf5ed9-fb25-4f7e-8bd3-9160af631452     |0               |10
Factory_Default_1_TLOC_Branch_Template                          |vedge-C8000V       |417e25b1-c37e-4ca9-8e06-fe048c76de9d     |0               |17
Default_Azure_vWAN_C8000V_Template_V01                          |vedge-C8000V       |0b2c653b-d8b2-4c21-8503-ce23f3017bb0     |0               |11
Default_BOOTSTRAP_DHCP_8000V_Template_V01                       |vedge-C8000V       |7e91fe82-f218-4d22-b0b9-d82f584c66e8     |0               |11
Default_SDBranch_ISRv_Template_V01                              |vedge-ISRv         |666f1052-29c3-412e-8fc7-7831b3c72351     |0               |16
Default_MEGAPORT_ICGW_C8000V_Template_V01                       |vedge-C8000V       |cd31fcdc-5af0-4ae4-ad9d-f10aea13aed3     |0               |9
Default_EQUINIX_ICGW_C8000V_Template_V01                        |vedge-C8000V       |0339b4c1-27c4-4668-b6f5-0e504fa27c4f     |0               |10
Default_BOOTSTRAP_STATIC_8000V_Template_V01                     |vedge-C8000V       |7c15e801-6e04-4259-9c28-252f9a059e26     |0               |10
Default_EQUINIX_DHCP_DNS_ICGW_CSR1000V_Template_V02             |vedge-CSR-1000v    |b869b565-850d-42e4-ad83-100846414d08     |0               |11
Factory_Default_C8000V_V01                                      |vedge-C8000V       |89e94daa-24e1-4b98-81d0-6d55fb33e8a1     |0               |22
edge_basic                                                      |vedge-C8000V       |1fe9f5b1-9d6b-476c-9276-7ab896261554     |0               |16
controller_basic                                                |vsmart             |db1a522f-8f90-4a3a-adf3-fbd7ee920a29     |1               |8
```

Figure 8-15 *Output of the Python Script from Example 8-12*

This chapter has explored several Cisco solutions and their REST APIs. It has explained authentication and authorization methods, and you have obtained basic information from the APIs. This chapter has provided a basic introduction to these extensive APIs and the features that they expose. We encourage you to continue your exploration of these APIs and build your own use cases, automation, and network programmability projects.

Exam Preparation Tasks

As mentioned in the section "How to Use This Book" in the Introduction, you have a couple of choices for exam preparation: the exercises here, Chapter 19, "Final Preparation," and the exam simulation questions on the companion website.

Review All Key Topics

Review the most important topics in this chapter, noted with the Key Topic icon in the outer margin of the page. Table 8-2 lists these key topics and the page number on which each is found.

Table 8-2 *Key Topics*

Key Topic Element	Description	Page Number
List	Qualities of a good SDK	
List	Advantages of SDKs	
List	The Meraki cloud platform provides several APIs from a programmability perspective	
List	Cisco Catalyst Center REST APIs and SDKs	
List	Cisco Catalyst SD-WAN products that perform different functions	

Define Key Terms

Define the following key terms from this chapter and check your answers in the glossary:

software development kit (SDK)
Python Enhancement Proposals (PEP)
Bluetooth Low Energy (BLE)
MQ Telemetry Transport (MQTT)
Cisco Digital Network Architecture (DNA)
data as a service (DaaS)
Software Image Management (SWIM) API
Plug and Play (PnP) API
Cisco Software Defined-WAN (SD-WAN)
Open Shortest Path First (OSPF)
Enhanced Interior Gateway Routing Protocol (EIGRP)
Border Gateway Protocol (BGP)
Cisco Express Forwarding (CEF)
software-defined networking (SDN)
Overlay Management Protocol (OMP)

Chapter 9. Cisco Compute

This chapter covers the following topics:

- **Cisco ACI:** This section describes Cisco ACI and the APIs it exposes.
- **Cisco UCS Manager:** This section covers Cisco UCS Manager and the public APIs that come with it.
- **Cisco Intersight:** This section introduces Cisco Intersight and its REST API interface.

This chapter begins by exploring Cisco data center technologies and the software development kits (SDKs) and application programming interfaces (APIs) that are available with them. First, the chapter provides an introduction into the Cisco Application Centric Infrastructure (ACI), the components that make up the solution, how the Cisco ACI REST API looks, and what resources are exposed over the API. It also covers a popular Python library called acitoolkit to extract data from the API. Following that, the chapter looks into the Cisco Unified Computing System (UCS) and how all the components of the solution work together to offer one of the most comprehensive and scalable data center compute solutions available today. In addition, the chapter includes an overview of Cisco UCS Manager and the XML API it provides and discusses how to interact with this API via **curl** commands and the Cisco UCS Manager SDK. Finally, the chapter ends with coverage of Cisco Intersight, a Software as a Service (SaaS) product that takes Cisco UCS management into the cloud. The chapter wraps up with a discussion of the REST API interface of Cisco Intersight and the Python SDK that come with it.

"Do I Know This Already?" Quiz

The "Do I Know This Already?" quiz allows you to assess whether you should read this entire chapter thoroughly or jump to the "Exam Preparation Tasks" section. If you are in doubt about your answers to these questions or your own assessment of your knowledge of the topics, read the entire chapter. Table 9-1 lists the major headings in this chapter and their corresponding "Do I Know This Already?" quiz questions. You can find the answers in Appendix A, "Answers to the 'Do I Know This Already?' Quizzes."

Table 9-1 *"Do I Know This Already?" Section-to-Question Mapping*

Foundation Topics Section	Questions
Cisco ACI	1–3
Cisco UCS Manager	4–6
Cisco Intersight	7–8

Caution:

The goal of self-assessment is to gauge your mastery of the topics in this chapter. If you do not know the answer to a question or are only partially sure of the answer, you should mark that question as wrong for purposes of the self-assessment. Giving yourself credit for an answer you correctly guess skews your self-assessment results and might provide you with a false sense of security.

1. On what family of switches is Cisco ACI running?
   - **a.** Cisco Catalyst 9000
   - **b.** Cisco Nexus 9000
   - **c.** Cisco Nexus 7000
   - **d.** Cisco Catalyst 6800
2. An ACI bridge domain can be associated with multiple VRFs.
   - **a.** True
   - **b.** False
3. What is the Cisco ACI REST API endpoint used for authentication?
   - **a.** https://APIC_IP_or_Hostname/api/aaaLogin.json

b. https://APIC_IP_or_Hostname/api/login

c. https://APIC_IP_or_Hostname/api/v1/aaaLogin

d. https://APIC_IP_or_Hostname/api/v1/login.json

4. Within Cisco UCS Manager, what is the logical construct that contains the complete configuration of a physical server?

   a. Server profile

   b. Service profile

   c. Template profile

   d. None of these answers are correct.

5. What is the Cisco UCS Manager Python SDK library called?

   a. ucsmsdk

   b. ucssdk

   c. ucsm

   d. ciscoucsm

6. What is the managed object browser called in Cisco UCS Manager?

   a. mobrowser

   b. ucsmobrowser

   c. visore

   d. ucsvisore

7. How are the managed objects organized in Cisco Intersight?

   a. Management Information Table

   b. Hierarchical Management Information Tree

   c. Management Information Model

   d. Hierarchical Managed Objects Model

8. What does the Cisco Intersight REST API key contain?

   a. keyId and keySecret

   b. token

   c. accessKey and secretKey

   d. cookie

Foundation Topics

Cisco ACI

Cisco *Application Centric Infrastructure (ACI)* is the SDN-based solution from Cisco for data center deployment, management, and monitoring. The solution is based on two components: the Cisco Nexus family of switches and Cisco *Application Policy Infrastructure Controller (APIC)*.

The Cisco Nexus 9000 family of switches can run in two separate modes of operation based on the software that is loaded on them. The first mode is called standalone, or NX-OS, mode, which means the switches act like regular Layer 2/Layer 3 data center devices that usually are managed individually. In the second mode, called ACI mode, the Cisco Nexus devices are part of an ACI fabric and are managed in a centralized fashion.

The central controller for the ACI fabric is the Cisco Application Policy Infrastructure Controller. This controller is the main architectural component of the Cisco ACI solution and provides a single point of automation and management for the Cisco ACI fabric, policy enforcement, and health monitoring. The Cisco APIC was built on an API-first architecture from its inception. On top of this API, a command-line interface (CLI) and a graphical user interface (GUI) have been developed. The API is exposed through a REST interface and is accessible as a northbound interface so that users and developers can integrate and develop their own custom solutions on top of the Cisco APIC and Cisco ACI fabric.

The Cisco APIC interacts and manages the Cisco Nexus switches through the OpFlex protocol that is exposed as a southbound interface. From an SDN controller perspective, similar to the Cisco Catalyst Center controller in Chapter 8, "Cisco Enterprise Networking Management Platforms and APIs," a northbound interface specifies the collection of protocols that a user can use to interact and program the controller, whereas a southbound interface specifies the protocols and interfaces that the controller uses to interact with the devices it manages.

The Cisco APIC has the following features and capabilities:

- Application-centric network policy engine for physical, virtual, and cloud infrastructure
- Data model–based declarative provisioning
- Design built around open standards and open APIs
- Cisco ACI fabric inventory and configuration
- Software image management
- Fault, event, and performance monitoring and management
- Integration with third-party management systems such as VMware, Microsoft, and OpenStack
- Cloud APIC appliance for Cisco Cloud ACI deployments in public cloud environments

At a minimum, three APICs in a cluster are needed for high availability.

The Cisco ACI fabric is built in a leaf/spine architecture. As the name implies, some of the Cisco Nexus switches that are part of the ACI fabric are called *leaves* and perform a function similar to that of an access switch, to which both physical and virtual endpoint servers are connected. Some of the switches are called *spines* and perform a function similar to that of a distribution switch to which all the access switches are connected. Figure 9-1 shows how all the Cisco ACI fabric components come together.

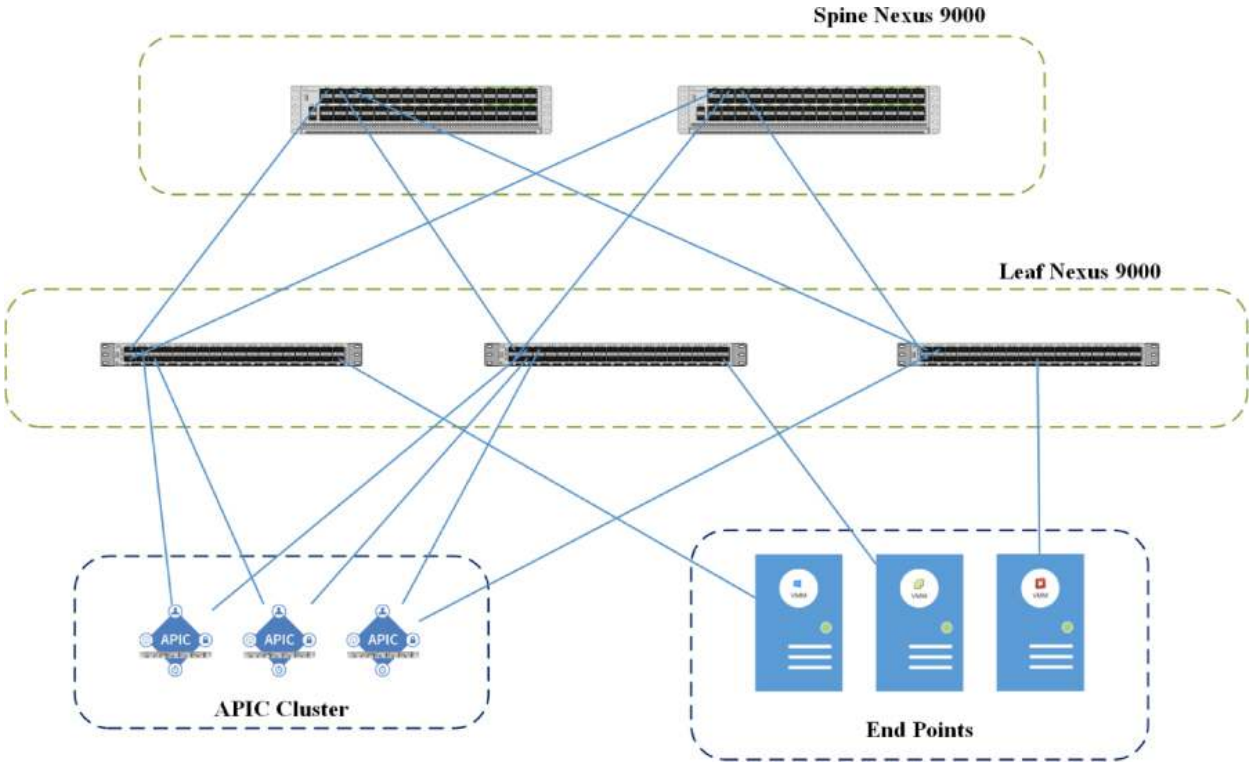


Figure 9-1 *Cisco ACI Fabric Architecture*

It is important to choose the right switches for the right function because not all Cisco Nexus 9000 switches support all functions in a leaf/spine architecture. The leaf switches connect to all the spine switches and to endpoint devices, including the Cisco APICs. The Cisco APICs will never connect to spine switches. Spine switches can connect only to leaf switches and will never interconnect with each other. The ACI fabric provides consistent low-latency forwarding across high-bandwidth links (40 Gbps, 100 Gbps, and 400 Gbps). Data traffic that has the source and destination on the same leaf switch is handled locally. When the traffic source and destination are on separate leaf switches, they are always only one spine switch away. The whole ACI fabric operates as a single Layer 3 switch, so between a data traffic source and destination, it will always be at most one Layer 3 hop.

The configuration of the ACI fabric is stored in the APIC using an object-oriented schema. This configuration represents the logical model of the fabric. The APIC compiles the logical model and renders the policies into a concrete model that runs in the physical infrastructure. Figure 9-2 shows the relationship between the logical model, the concrete model, and the operating system running on the switches.

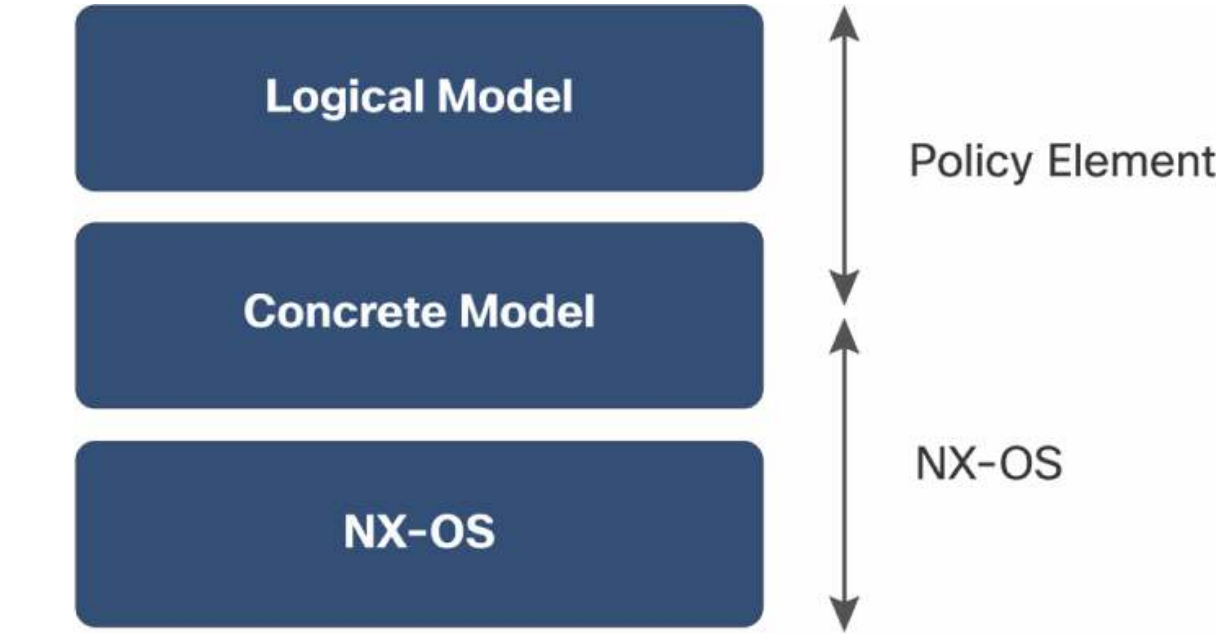


Figure 9-2 *Relationship Between the Logical Model, the Concrete Model, and the Operating System*

All the switches contain a complete copy of the concrete model. When a policy that represents a configuration is created in the APIC, the controller updates the logical model. It then performs the intermediate step of creating a complete policy that it pushes into all the switches where the concrete model is updated. The Cisco Nexus 9000 switches can execute the concrete model only when running in ACI mode. Each switch has a copy of the concrete model. If the APIC goes offline, the fabric keeps functioning, but modifications to the fabric policies are not possible.

The ACI policy model enables the specification of application requirements. When a change is initiated to an object in the fabric, the APIC first applies that change to the policy model. This policy model change triggers a change to the concrete model and the actual managed endpoint. This management framework is called the *model-driven framework*. In this model, the system administrator defines the desired state of the fabric but leaves the implementation up to the APIC. This means that the data center infrastructure is no longer managed in isolated, individual component configurations but holistically, enabling automation and flexible workload provisioning. In this type of infrastructure, network-attached services can be deployed easily because the APIC provides an automation framework to manage the complete lifecycle of these services. As workloads move and changes happen, the controller reconfigures the underlying infrastructure to ensure the policies are still in place for the end hosts.

The Cisco ACI fabric is composed of physical and logical components. These components are recorded in the *Management Information Model (MIM)* and can be represented in a hierarchical *management information tree (MIT)*. Each node in the MIT represents a *managed object (MO)*. An MO can represent a concrete object (such as a switch, adapter, or power supply) or a logical object (such as an application profile, *endpoint group [EPG]*, or error message). All components of the ACI fabric can be represented as managed objects. Figure 9-3 provides an overview of the MIT and its elements.

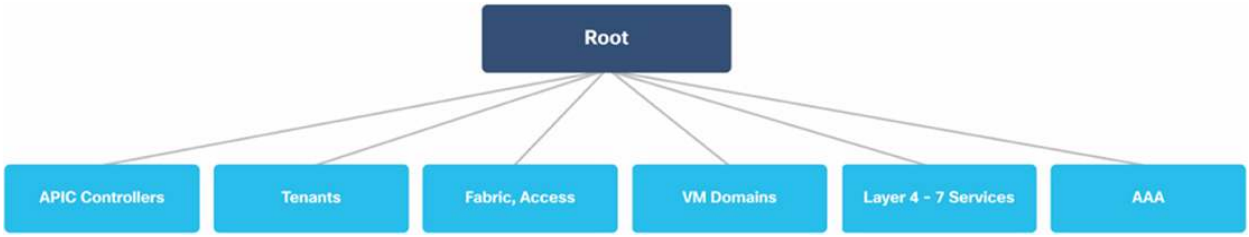


Figure 9-3 *Cisco ACI Management Information Tree*

The MIT hierarchical structure starts at the top with the root object and contains parent and child nodes. Each node in the tree is an MO, and each object in the fabric has a *distinguished name (DN)*. The DN describes the object and specifies its location in the tree. The following managed objects contain the policies that control the operation of the fabric:

- APICs are the clustered fabric controllers that provide management, application, and policy deployment for the fabric.
- Tenants represent containers for policies that are grouped for a specific access domain. The following four kinds of tenants are currently supported by the system:
  - User tenants are needed by the fabric administrator to cater to the needs of the fabric users.
  - The common tenant is provided by the system and can be configured by the fabric administrator. This tenant contains policies and resources that can be shared by all tenants. Examples of such resources are firewalls, load balancers, and intrusion detection systems.
  - The infra tenant is provided by the system and can be configured by the fabric administrator. It contains policies that manage the operation of infrastructure resources.
  - The management tenant is provided by the system and can be configured by the fabric administrator. This tenant contains policies and resources used for in-band and out-of-band configuration of fabric nodes.
- Access policies control the operation of leaf switch access ports that provide fabric connectivity to resources such as virtual machine (VM) hypervisors, compute, and storage devices. Several access policies come built in with the ACI fabric by default. The fabric administrator can tweak these policies or create new ones as necessary.
- Fabric policies control the operation of the switch fabric ports. Configurations for time synchronization, routing protocols, and domain name resolution are managed with these policies.
- Virtual machine domains group VM controllers that require similar networking policy configurations. The APIC communicates with the VM controller to push network configurations all the way to the virtual machine level.
- The Layer 4 to Layer 7 service integration automation framework enables the system to respond to services coming online or going offline.
- Access, authentication, and accounting (AAA) policies control user privileges, roles, and security domains for the ACI fabric.

The hierarchical policy model fits well with the REST API interface. As the ACI fabric performs its functions, the API reads and writes to objects in the MIT. The API resources represented by URLs map directly into the distinguished names that identify objects in the MIT.

Next, let's explore the building blocks of the Cisco ACI fabric policies. Tenants are top-level MOs that identify and separate administrative controls, application policies, and failure domains. A tenant can represent a customer in a managed service provider environment, an organization in an enterprise environment, or just a convenient grouping of objects and policies. A tenant's sublevel objects can be grouped into two categories: tenant networking and tenant policy. Figure 9-4 illustrates how a tenant is organized and the main networking and policy components.

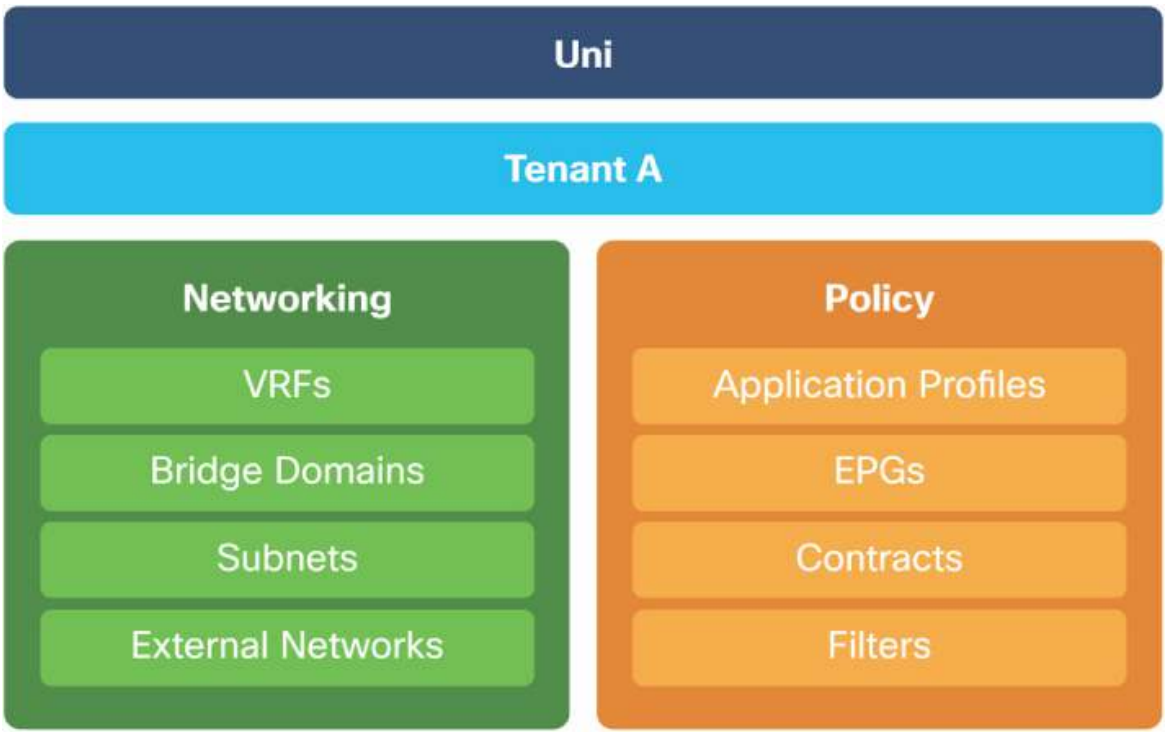


Figure 9-4 *Cisco ACI Tenant Components*

The tenant networking objects provide Layer 2 and Layer 3 connectivity between the endpoints and consist of the following constructs: *virtual routing and forwarding (VRF) instances*, bridge domains, subnets, and external networks. Figure 9-5 displays in more detail how the tenant networking constructs are organized.

Figure 9-5 *Cisco ACI Tenant Networking Components*

VRFs, also called *contexts* and *private networks*, are isolated routing tables. A VRF defines a Layer 3 address domain. A tenant can contain one or multiple VRFs. VRFs exist on any leaf switch that has a host assigned to the VRF. All the endpoints within a Layer 3 domain must have unique IP addresses because traffic can flow between these devices if the policy allows.

Bridge domains represent the Layer 2 forwarding domains within the fabric and define the unique MAC address space and flooding domain for broadcast, unknown unicast, and multicast frames. Each bridge domain is associated with only one VRF; however, a VRF can be associated with multiple bridge domains. Bridge domains can contain multiple subnets; this capability is different from regular VLANs, which are usually associated with only one subnet.

Subnets are the Layer 3 networks that provide IP address space and gateway services for endpoints to be able to connect to the network. Each subnet is associated with only one bridge domain. Subnets can be

- **Public:** The subnet can be exported to a routed connection.
- **Private:** The subnet is confined within its tenant.
- **Shared:** The subnet can be shared and exposed in multiple VRFs in the same tenant or across tenants as part of a shared service.

External bridged networks connect the ACI fabric to legacy Layer 2/Spanning Tree Protocol networks. This connection is usually needed as part of the migration process from a traditional network infrastructure to an ACI network.

External routed networks create a Layer 3 connection with a network outside the ACI fabric. Layer 3 external routed networks can be configured using static routes or routing protocols like BGP, OSPF, and EIGRP.

Tenant policy objects are focused on the policies and services that the endpoints receive. A tenant policy consists of application profiles, endpoint groups, contracts, and filters. Figure 9-6 displays how tenant policy objects, application profiles, and EPGs are

organized within different bridge domains.

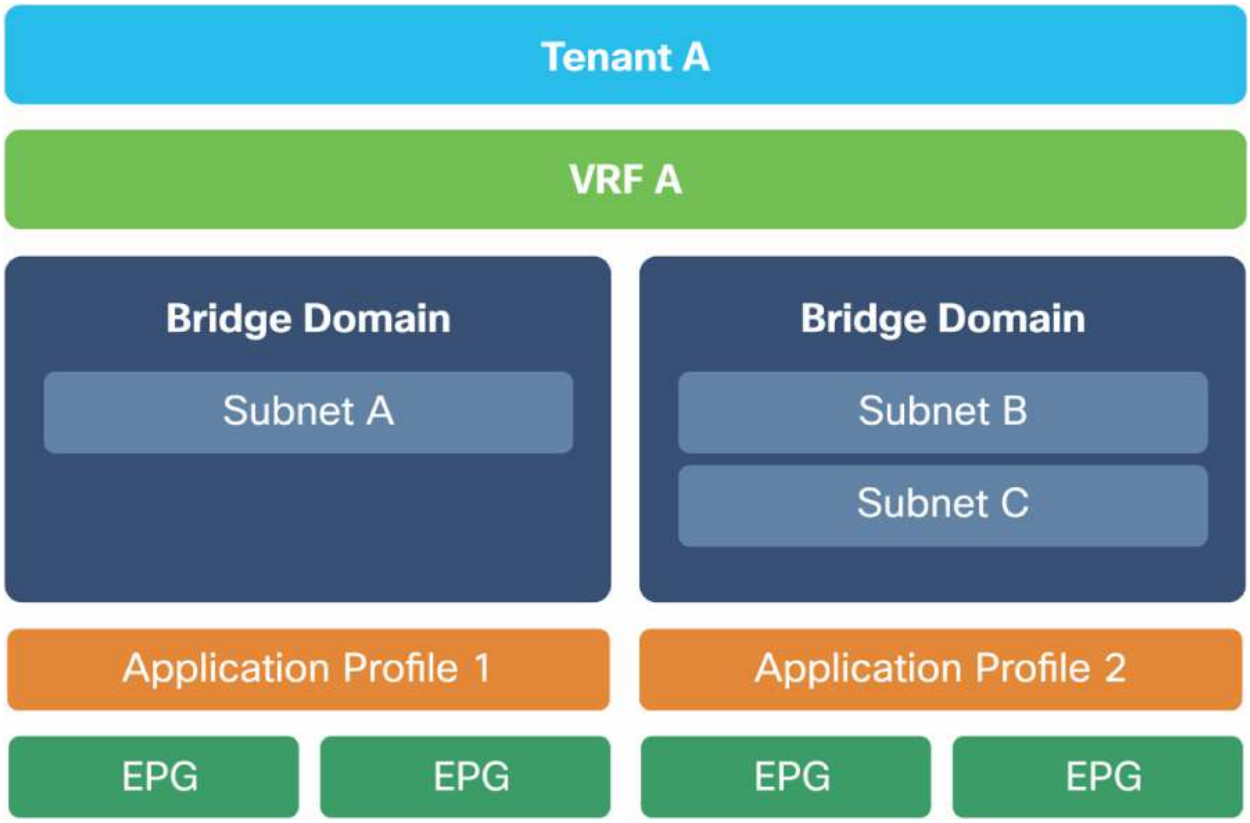


Figure 9-6 *Cisco ACI Tenant Policy Components*

An application profile defines the policies, services, and relationships between endpoint groups. Application profiles contain one or more EPGs. Applications typically contain multiple components, such as a web-based front end, an application logic layer, and one or more databases in the back end. The application profile contains as many EPGs as necessary, and these EPGs are logically related to providing the capabilities of the application.

The endpoint group is the most important object in the policy model. An EGP is a collection of endpoints that have common policy requirements, such as security, virtual machine mobility, quality of service (QoS), or Layer 4 to Layer 7 services. In the Cisco ACI fabric, endpoints have an identity represented by their address, a location, and attributes, and they can be physical or virtual. Endpoint examples include servers, virtual machines, clients on the Internet, or network-attached storage devices. Rather than configure and manage endpoints individually, you can place them in EPGs and manage them as a group. Policies apply to EPGs and never to individual endpoints. Each EPG can be related to only one bridge domain.

Contracts define the policies and services that get applied to EPGs. Contracts can be used for redirecting service to a Layer 4 to Layer 7 device, assigning QoS values, and controlling the traffic flow between EPGs. EPGs can communicate only with other EPGs based on contract rules. Contracts specify the protocols and ports allowed between EPGs. If there is no contract, inter-EPG communication is disabled by default. For intra-EPG communication, no contract is required because this traffic is always allowed by default. The relationship between an EPG and a contract can be either a consumer or provider. EPG providers expose contracts with which a consumer EPG must comply. When an EPG consumes a contract, the endpoints in the consuming EPG can initiate communication with any endpoint from the provider EPG. Figure 9-7 displays this contractual relationship between providing and consuming EPGs.

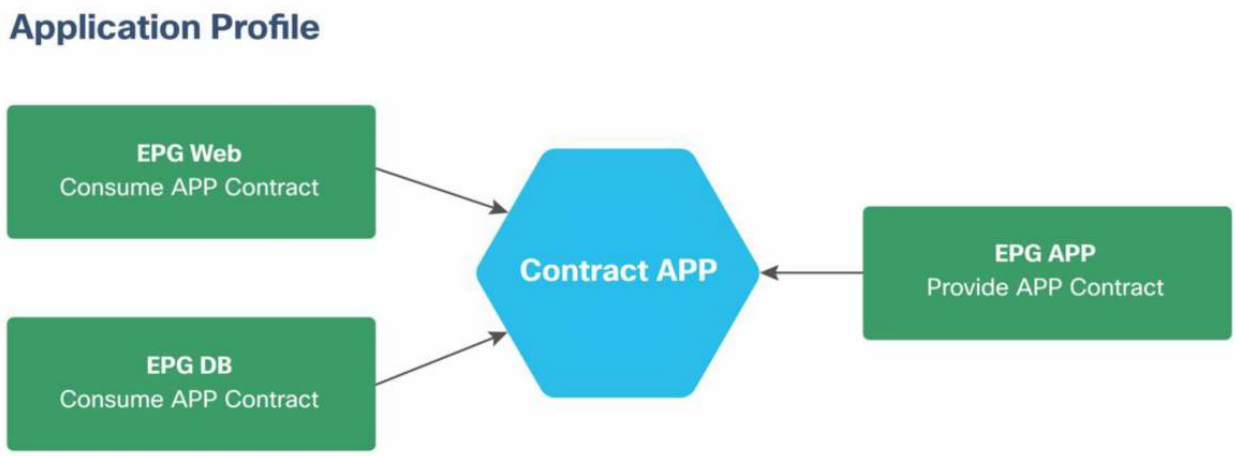


Figure 9-7 *Cisco ACI Application Profiles and Contracts*

Filters are the objects that define protocols and port numbers that are used in contracts. Filter objects can contain multiple protocols and ports, and contracts can consume multiple filters.

As mentioned previously, the APIC REST API is a programmatic interface that uses the REST architecture. The API accepts and returns HTTP or HTTPS messages that contain JSON or XML documents. Any programming language can be used to generate the messages and the JSON or XML documents that contain the API methods or managed object (MO) attributes. Whenever information is retrieved and displayed, it is read through the REST API, and whenever configuration changes are done, they are written through the REST API. The REST API also provides a way of subscribing to push-based event notification; consequently, when a change occurs in the MIT, an event can be sent through a web socket.

A generic APIC REST API uniform resource identifier (URI) looks like the following:

https://APIC_Host:port/api/{mo|class}/{dn|classname}.{xml|json}?[options]

Since the REST API matches one to one with the MIT, defining the URI to access a certain resource is important. First, it defines the protocol (http or https) and the hostname or IP address of the APIC instance. Next, the /api indicates that the API is invoked. After that, the next part of the URI specifies whether the operation will be for an MO or class. The next component defines either the fully qualified domain name (DN) for MO-based queries or the class name for class-based queries. The final mandatory part of the request is the encoding format, which can be either XML or JSON. The APIC ignores Content-Type and other headers, so the method just explained is the only one accepted.

APIC REST API username- and password-based authentication uses a special URI, including aaaLogin, aaaLogout, and aaaRefresh, as the DN targets of a POST operation. The payloads contain a simple XML or JSON document containing the MO representation of an aaaUser object.

The following examples use the Cisco DevNet always-on APIC instance available at https://sandboxapicdc.cisco.com with a username value of admin and a password of !v3G@!4@Y to interact with an ACI fabric using the APIC REST API interface. When you use **curl**, the authentication API call should look like the one shown in Example 9-1.

Example 9-1 ***curl*** *Command for Cisco APIC Authentication*

```
curl -k -X POST \
  https://sandboxapicdc.cisco.com/api/aaaLogin.json \
  -d '{
  "aaaUser" : {
    "attributes" : {
      "name" : "admin",
      "pwd" : "!v3G@!4@Y"
    }
  }
}'
```

The information returned from the APIC should look like the response in Example 9-2.

Example 9-2 *Cisco APIC Authentication Response*

```
{
    "totalCount": "1",
    "imdata": [
        {
            "aaaLogin": {
                "attributes": {
                    "token":
"eyJhbGciOiJSUzI1NiIsImtpZCI6ImM1ODRnb2VlYTh4bG1saGRnaWZkcHk3YmQ5NWl4b3U4IiwidHlwIjoiand0In0.eyJyYmFjI
jpbeyJkb21haW4iOiJhbGwiLCJyb2xlc1IiOjAsInJvbGVzVyI6MX1dLCJpc3MiOiJBQ0kgQVBJQyIsInVzZXJuYW1lIjoiYWRtaW4
iLCJ1c2VyaWQiOjE1Mzc0LCJ1c2VyZmxhZ3MiOjAsImlhdCI6MTczNjE0MDgyNCwiZXhwIjoxNzM2MTQxNDI0LCJzZXNzaW9uaWQiO
iJtWnduenBZTVNJYVd5ZDA0MXBXTGZBPT0ifQ.CCn8YBwG4HY9nSigIZcyGycAIIC1mBYbiSdvLhaZBEZK-
z0IFagt_LSJ0oFcQBti1OAvAxuNSOc9-N_gcfYKZqbMbfhgeWH2Fx3-
MEoxC6teVt3o3xW265tXmlL9GnnW1ES1rGONKNiHN2eT1PD3lOJS2gdfZoqQAkZHojDPN-MWj8oVFn8qMtuiX2LT-
fkgpYbbvgO4McNIxSqOaurbUPYD_v6Pz02fU1iEVxF4rCHCLPKt5qTpsS4TEMuD6JiEr7xzS-
mgywASk_BeJu0tzyMaKESPlr4KcEflYli8P8iZCXMFVLxUNl4SbsGd8Zd4bXBZueZrmrZn0gosW-kVWg",
                    "siteFingerprint": "c584goeea8xlmlhdgifdpy7bd95ixou8",
                    "refreshTimeoutSeconds": "600",
                    "maximumLifetimeSeconds": "86400",
                    "guiIdleTimeoutSeconds": "65000",
                    "restTimeoutSeconds": "90",
                    "creationTime": "1736140824",
                    "firstLoginTime": "1736140824",
                    "userName": "admin",
                    "remoteUser": "false",
                    "unixUserId": "15374",
                    "sessionId": "mZwnzpYMSIaWyd041pWLfA==",
                    "lastName": "",
                    "firstName": "",
                    "changePassword": "no",
                    "version": "6.0(2h)",
                    "buildTime": "Sun Feb 26 22:50:06 PST 2023",
                    "node": "topology/pod-1/node-1"
}
…omitted output
}
```


The response to the POST operation contains an authentication token that will be used in subsequent API operations as a cookie named APIC-cookie.

Next, let's get a list of all the ACI fabrics that are managed by this APIC instance. The URI for this GET operation is https://sandboxapicdc.cisco.com/api/node/class/fabricPod.json; the APIC-cookie is specified for authentication purposes. The **curl** request should look similar to the one shown in Example 9-3.

Example 9-3 ***curl*** *Command to Get a List of All ACI Fabrics*

```
curl -k -X GET \
  https://sandboxapicdc.cisco.com/api/node/class/fabricPod.json \
  -H 'Cookie: APIC-cookie=
eyJhbGciOiJSUzI1NiIsImtpZCI6ImM1ODRnb2VlYTh4bG1saGRnaWZkcHk3YmQ5NWl4b3U4IiwidHlwIjoiand0In0.eyJyYmFjIj
pbeyJkb21haW4iOiJhbGwiLCJyb2xlc1IiOjAsInJvbGVzVyI6MX1dLCJpc3MiOiJBQ0kgQVBJQyIsInVzZXJuYW1lIjoiYWRtaW4i
LCJ1c2VyaWQiOjE1Mzc0LCJ1c2VyZmxhZ3MiOjAsImlhdCI6MTczNjE0MDgyNCwiZXhwIjoxNzM2MTQxNDI0LCJzZXNzaW9uaWQiOi
JtWnduenBZTVNJYVd5ZDA0MXBXTGZBPT0ifQ.CCn8YBwG4HY9nSigIZcyGycAIIC1mBYbiSdvLhaZBEZK-
z0IFagt_LSJ0oFcQBti1OAvAxuNSOc9-N_gcfYKZqbMbfhgeWH2Fx3-
MEoxC6teVt3o3xW265tXmlL9GnnW1ES1rGONKNiHN2eT1PD3lOJS2gdfZoqQAkZHojDPN-MWj8oVFn8qMtuiX2LT-
fkgpYbbvgO4McNIxSqOaurbUPYD_v6Pz02fU1iEVxF4rCHCLPKt5qTpsS4TEMuD6JiEr7xzS-
mgywASk_BeJu0tzyMaKESPlr4KcEflYli8P8iZCXMFVLxUNl4SbsGd8Zd4bXBZueZrmrZn0gosW-kVWg'
```

The response received from this instance of APIC should look like the one in Example 9-4.

Example 9-4 ***curl*** *Command Response with a List of ALL ACI Fabrics*

```
{
    "totalCount": "1",
    "imdata": [
        {
            "fabricPod": {
                "attributes": {
                    "childAction": "",
                    "dn": "topology/pod-1",
                    "id": "1",
                    "lcOwn": "local",
                    "modTs": "2025-01-03T07:06:50.175+00:00",
                    "monPolDn": "uni/fabric/monfab-default",
                    "podType": "physical",
                    "status": ""
                }
            }
        }
    ]
}
```


From this response, you can see that the always-on Cisco DevNet Sandbox APIC instance manages only one ACI fabric called pod-1.

Next, let's find out more information about pod-1 and discover how many devices are part of this fabric. The API URI link for the resource that will return this information is https://sandboxapicdc.cisco.com/api/node/class/topology/pod-1/topSystem.json. You can specify the APIC-cookie again, and the GET request should look like the one shown in Example 9-5.

Example 9-5 ***curl*** *Command to Get ACI Pod Information*

```
curl -k -X GET \
  https://sandboxapicdc.cisco.com/api/node/class/topology/pod-1/topSystem.json \
  -H 'Cookie: APIC-
cookie=eyJhbGciOiJSUzI1NiIsImtpZCI6ImM1ODRnb2VlYTh4bG1saGRnaWZkcHk3YmQ5NWl4b3U4IiwidHlwIjoiand0In0.eyJ
yYmFjIjpbeyJkb21haW4iOiJhbGwiLCJyb2xlc1IiOjAsInJvbGVzVyI6MX1dLCJpc3MiOiJBQ0kgQVBJQyIsInVzZXJuYW1lIjoiY
WRtaW4iLCJ1c2VyaWQiOjE1Mzc0LCJ1c2VyZmxhZ3MiOjAsImlhdCI6MTczNjE0MDgyNCwiZXhwIjoxNzM2MTQxNDI0LCJzZXNzaW9
uaWQiOiJtWnduenBZTVNJYVd5ZDA0MXBXTGZBPT0ifQ.CCn8YBwG4HY9nSigIZcyGycAIIC1mBYbiSdvLhaZBEZK-
z0IFagt_LSJ0oFcQBti1OAvAxuNSOc9-N_gcfYKZqbMbfhgeWH2Fx3-
MEoxC6teVt3o3xW265tXmlL9GnnW1ES1rGONKNiHN2eT1PD3lOJS2gdfZoqQAkZHojDPN-MWj8oVFn8qMtuiX2LT-
fkgpYbbvgO4McNIxSqOaurbUPYD_v6Pz02fU1iEVxF4rCHCLPKt5qTpsS4TEMuD6JiEr7xzS-
mgywASk_BeJu0tzyMaKESPlr4KcEflYli8P8iZCXMFVLxUNl4SbsGd8Zd4bXBZueZrmrZn0gosW-kVWg'
```

The redacted response from APIC should look similar to the one shown in Example 9-6.

Example 9-6 *REST API Response Containing Details About the Cisco ACI Pod*

```
{
    "totalCount": "4",
    "imdata": [
        {
            "topSystem": {
                "attributes": {
                    "address": "10.0.0.1",
                    "bootstrapState": "none",
                    "childAction": "",
                    "clusterTimeDiff": "0",
                    "configIssues": "",
                    "controlPlaneMTU": "9000",
                    "currentTime": "2025-01-06T05:33:18.119+00:00",
                    "dn": "topology/pod-1/node-1/sys",
                    "enforceSubnetCheck": "no",
                    "etepAddr": "0.0.0.0",
                    "fabricDomain": "ACI Fabric1",
                    "fabricId": "1",
                    "fabricMAC": "00:22:BD:F8:19:FF",
                    "id": "1",
                    "inbMgmtAddr": "192.168.11.1",
                    "inbMgmtAddr6": "fc00::1",
                    "inbMgmtAddr6Mask": "0",
                    "inbMgmtAddrMask": "24",
                    "inbMgmtGateway": "192.168.11.254",
                    "inbMgmtGateway6": "::",
                    "lastRebootTime": "2025-01-03T06:58:50.230+00:00",
                    "lastResetReason": "unknown",
                    "lcOwn": "local",
                    "modTs": "2025-01-05T06:00:43.978+00:00",
                    "mode": "unspecified",
                    "monPolDn": "uni/fabric/monfab-default",
                    "name": "apic1",
                    "nameAlias": "",
                    "nodeType": "unspecified",
                    "oobMgmtAddr": "10.10.20.14",
                    "oobMgmtAddr6": "fe80::200:ff:fe0:0",
                    "oobMgmtAddr6Mask": "0",
                    "oobMgmtAddrMask": "24",
                    "oobMgmtGateway": "10.10.20.254",
                    "oobMgmtGateway6": "2001:420:28e:2020:acc:68ff:fe28:b540",
                    "podId": "1",
                    "remoteNetworkId": "0",
                    "remoteNode": "no",
                    "rlOperPodId": "0",
                    "rlRoutableMode": "no",
                    "rldirectMode": "no",
                    "role": "controller",
                    "serial": "TEP-1-1",
                    "serverType": "unspecified",
                    "siteId": "0",
                    "state": "in-service",
                    "status": "",
                    "systemUpTime": "02:22:34:28.000",
                    "tepPool": "0.0.0.0",
                    "unicastXrEpLearnDisable": "no",
                    "version": "6.0(2h)",
                    "virtualMode": "no"
                }
            }
        },
```

```
{
    "topSystem": {
        "attributes": {
            "address": "10.0.56.64",
            "bootstrapState": "done",
            "childAction": "",
            "clusterTimeDiff": "-9",
            "configIssues": "",
            "controlPlaneMTU": "9000",
            "currentTime": "2025-01-06T05:33:18.128+00:00",
            "dn": "topology/pod-1/node-101/sys",
            "enforceSubnetCheck": "yes",
            "etepAddr": "0.0.0.0",
            "fabricDomain": "ACI Fabric1",
            "fabricId": "1",
            "fabricMAC": "00:22:BD:F8:19:FF",
            "id": "101",
            "inbMgmtAddr": "0.0.0.0",
            "inbMgmtAddr6": "::",
            "inbMgmtAddr6Mask": "0",
            "inbMgmtAddrMask": "0",
            "inbMgmtGateway": "0.0.0.0",
            "inbMgmtGateway6": "::",
            "lastRebootTime": "2025-01-03T06:58:50.069+00:00",
            "lastResetReason": "unknown",
            "lcOwn": "local",
            "modTs": "2025-01-05T16:57:16.182+00:00",
            "mode": "unspecified",
            "monPolDn": "uni/fabric/monfab-default",
            "name": "Leaf_101",
            "nameAlias": "",
            "nodeType": "unspecified",
            "oobMgmtAddr": "192.168.1.10",
            "oobMgmtAddr6": "::",
            "oobMgmtAddr6Mask": "32",
            "oobMgmtAddrMask": "24",
            "oobMgmtGateway": "192.168.1.100",
            "oobMgmtGateway6": "::",
            "podId": "1",
            "remoteNetworkId": "0",
            "remoteNode": "no",
            "rlOperPodId": "1",
            "rlRoutableMode": "no",
            "rldirectMode": "no",
            "role": "leaf",
            "serial": "TEP-1-101",
            "serverType": "unspecified",
            "siteId": "0",
            "state": "in-service",
            "status": "",
            "systemUpTime": "02:22:34:28.000",
            "tepPool": "10.0.0.0/16",
            "unicastXrEpLearnDisable": "no",
            "version": "simsw-6.0(2h)",
            "virtualMode": "no"
        }
    }
},
{
    "topSystem": {
        "attributes": {
```

```
                "address": "10.0.56.66",
                "bootstrapState": "done",
                "childAction": "",
                "clusterTimeDiff": "-9",
                "configIssues": "",
                "controlPlaneMTU": "9000",
                "currentTime": "2025-01-06T05:33:18.129+00:00",
                "dn": "topology/pod-1/node-102/sys",
                "enforceSubnetCheck": "yes",
                "etepAddr": "0.0.0.0",
                "fabricDomain": "ACI Fabric1",
                "fabricId": "1",
                "fabricMAC": "00:22:BD:F8:19:FF",
                "id": "102",
                "inbMgmtAddr": "0.0.0.0",
                "inbMgmtAddr6": "::",
                "inbMgmtAddr6Mask": "0",
                "inbMgmtAddrMask": "0",
                "inbMgmtGateway": "0.0.0.0",
                "inbMgmtGateway6": "::",
                "lastRebootTime": "2025-01-03T06:58:50.753+00:00",
                "lastResetReason": "unknown",
                "lcOwn": "local",
                "modTs": "2025-01-05T16:57:16.237+00:00",
                "mode": "unspecified",
                "monPolDn": "uni/fabric/monfab-default",
                "name": "Leaf_102",
                "nameAlias": "",
                "nodeType": "unspecified",
                "oobMgmtAddr": "192.168.1.9",
                "oobMgmtAddr6": "::",
                "oobMgmtAddr6Mask": "32",
                "oobMgmtAddrMask": "24",
                "oobMgmtGateway": "192.168.1.100",
                "oobMgmtGateway6": "::",
                "podId": "1",
                "remoteNetworkId": "0",
                "remoteNode": "no",
                "rlOperPodId": "1",
                "rlRoutableMode": "no",
                "rldirectMode": "no",
                "role": "leaf",
                "serial": "TEP-1-102",
                "serverType": "unspecified",
                "siteId": "0",
                "state": "in-service",
                "status": "",
                "systemUpTime": "02:22:34:28.000",
                "tepPool": "10.0.0.0/16",
                "unicastXrEpLearnDisable": "no",
                "version": "simsw-6.0(2h)",
                "virtualMode": "no"
            }
        }
    },
    {
        "topSystem": {
            "attributes": {
                "address": "10.0.56.65",
                "bootstrapState": "done",
                "childAction": "",
```

```
                    "clusterTimeDiff": "-14",
                    "configIssues": "",
                    "controlPlaneMTU": "9000",
                    "currentTime": "2025-01-06T05:33:18.132+00:00",
                    "dn": "topology/pod-1/node-201/sys",
                    "enforceSubnetCheck": "no",
                    "etepAddr": "0.0.0.0",
                    "fabricDomain": "ACI Fabric1",
                    "fabricId": "1",
                    "fabricMAC": "00:22:BD:F8:19:FF",
                    "id": "201",
                    "inbMgmtAddr": "0.0.0.0",
                    "inbMgmtAddr6": "::",
                    "inbMgmtAddr6Mask": "0",
                    "inbMgmtAddrMask": "0",
                    "inbMgmtGateway": "0.0.0.0",
                    "inbMgmtGateway6": "::",
                    "lastRebootTime": "2025-01-03T06:58:50.475+00:00",
                    "lastResetReason": "unknown",
                    "lcOwn": "local",
                    "modTs": "2025-01-05T16:57:16.204+00:00",
                    "mode": "unspecified",
                    "monPolDn": "uni/fabric/monfab-default",
                    "name": "Spine_201",
                    "nameAlias": "",
                    "nodeType": "unspecified",
                    "oobMgmtAddr": "192.168.1.8",
                    "oobMgmtAddr6": "::",
                    "oobMgmtAddr6Mask": "32",
                    "oobMgmtAddrMask": "24",
                    "oobMgmtGateway": "192.168.1.100",
                    "oobMgmtGateway6": "::",
                    "podId": "1",
                    "remoteNetworkId": "0",
                    "remoteNode": "no",
                    "rlOperPodId": "1",
                    "rlRoutableMode": "yes",
                    "rldirectMode": "yes",
                    "role": "spine",
                    "serial": "TEP-1-103",
                    "serverType": "unspecified",
                    "siteId": "0",
                    "state": "in-service",
                    "status": "",
                    "systemUpTime": "02:22:34:28.000",
                    "tepPool": "10.0.0.0/16",
                    "unicastXrEpLearnDisable": "no",
                    "version": "simsw-6.0(2h)",
                    "virtualMode": "no"
                }
            }
        }
    ]
}
```


From the response, you can see that this ACI fabric is made up of four devices: an APIC, two leaf switches, and one spine switch. Extensive information is returned about each device in this response, but it was modified to extract and display just a subset of that information. You are encouraged to perform the same steps and explore the APIC REST API either using the Cisco DevNet sandbox resources or your own instance of APIC.

As of this writing, several tools and libraries are available for Cisco ACI automation. An ACI Python SDK called Cobra can be used for advanced development. For basic day-to-day configuration and monitoring tasks and for getting started with ACI automation, you can use a Python library called acitoolkit. The acitoolkit library exposes a subset of the APIC object model that covers the most common ACI workflows.

Next, let's use acitoolkit to build a Python script that retrieves all the endpoints from an ACI fabric. Additional information about each endpoint will be displayed, such as the EPGs in which they are members, application profiles that are applied to those EPGs, as well as tenant membership, encapsulation, and MAC and IP addresses. A Python script that uses acitoolkit and accomplishes these tasks might look like the one shown in Example 9-7.

Example 9-7 *acitoolkit Example*

```
#! /usr/bin/env python
import acitoolkit.acitoolkit as aci

apic_url = 'https://sandboxapicdc.cisco.com'
username = 'admin'
password = '!v3G@!4@Y'

# Login to APIC
session = aci.Session(apic_url, username, password)
resp = session.login()
if not resp.ok:
    print('Could not login to APIC')
    exit()

endpoints = aci.Endpoint.get(session)
print('{0:19s}{1:14s}{2:10s}{3:8s}{4:17s}{5:10s}'.format("MAC ADDRESS",\
     "IP ADDRESS", "ENCAP", "TENANT", "APP PROFILE", "EPG"))
print('-'*80)

for ep in endpoints:
    epg = ep.get_parent()
    app_profile = epg.get_parent()
    tenant = app_profile.get_parent()
    print('{0:19s}{1:14s}{2:10s}{3:8s}{4:17s}{5:10s}'.format(ep.mac,\
        ep.ip, ep.encap, tenant.name, app_profile.name, epg.name))
```

The acitoolkit library supports Python 3; this example uses version 0.4 of the library. The script tested successfully with Python 3.9.6. Acitoolkit was installed using the Python 3 package manager pip, by issuing **pip install acitoolkit** in a Python 3.9.6 virtual environment.

First, the acitoolkit library is imported and will be referenced in the script using the short name aci. Three variables are defined next: apic_url contains the URL for the APIC instance that will be queried (in this case, the Cisco DevNet always-on APIC sandbox is used), and the username and password contain the login credentials for the APIC instance. Next, using the Session method of the aci class, a connection is established with the APIC. The Session method takes as input the three variables defined previously: the APIC URL, username, and password. Next, the success of the login action is verified. If the response to the login action is not okay, a message is displayed to the console ("Could not login to APIC"), and the script execution ends. If the login is successful, all the endpoints in the ACI fabric instance are stored in the endpoints variable. This is done by using the GET method of the aci.Endpoint class and passing in the current session object. Next, the headers of the table with the information that will be extracted are displayed to the console. As mentioned previously, the MAC address, IP address, encapsulation, tenant, application profile, and EPG will be retrieved from the APIC for all the endpoints in the fabric. The for iterative loop will go over one endpoint at a time and, using the get_parent() method, will go up one level in the MIT. First, it will retrieve the parent MO of the endpoint, which is the EPG. Recall that all endpoints in an ACI fabric are organized in EPGs. The parent object for an endpoint is therefore the EPG in which the endpoint is a member. Going up one level in the MIT, the parent object of the EPG is the application profile, and going up one more level, the parent object of the application profile is the tenant object. The epg, app_profile, and tenant variables will contain the respective EPG, application profile, and tenant values for each endpoint in the fabric. The last line of code in the script displays the required information for each endpoint to the console. The output of the script should look similar to the content shown in Figure 9-8.

MAC ADDRESS	IP ADDRESS	ENCAP	TENANT	APP PROFILE	EPG
44:CD:BB:C0:00:00	10.193.101.3	vlan-201	Heroes	Save_The_Planet	app
45:CD:BB:C0:00:00	2222::66:3	vlan-201	Heroes	Save_The_Planet	app
45:CD:BB:C0:00:00	10.193.102.3	vlan-200	Heroes	Save_The_Planet	web
45:CD:BB:C0:00:00	2222::66:3	vlan-202	Heroes	Save_The_Planet	db
44:CD:BB:C0:00:00	2222::65:3	vlan-200	Heroes	Save_The_Planet	web
44:CD:BB:C0:00:00	10.193.101.3	vlan-202	Heroes	Save_The_Planet	db
46:CD:BB:C0:00:00	10.193.102.4	vlan-201	Heroes	Save_The_Planet	app
46:CD:BB:C0:00:00	2222::66:4	vlan-200	Heroes	Save_The_Planet	web
46:CD:BB:C0:00:00	2222::66:4	vlan-202	Heroes	Save_The_Planet	db
42:CD:BB:C0:00:00	10.193.101.1	vlan-126	SnV	Chaos	Database
43:CD:BB:C0:00:00	2222::66:1	vlan-126	SnV	Chaos	Database
42:CD:BB:C0:00:00	2222::65:1	vlan-125	SnV	Chaos	Web
43:CD:BB:C0:00:00	10.193.102.1	vlan-125	SnV	Chaos	Web
43:CD:BB:C0:00:00	10.193.102.1	vlan-128	SnV	Power_Up	Database
43:CD:BB:C0:00:00	2222::66:1	vlan-121	SnV	Evolution_X	Web
43:CD:BB:C0:00:00	2222::66:1	vlan-127	SnV	Power_Up	Web
42:CD:BB:C0:00:00	10.193.101.1	vlan-121	SnV	Evolution_X	Web
42:CD:BB:C0:00:00	2222::65:1	vlan-124	SnV	Rescue	Database
43:CD:BB:C0:00:00	10.193.102.1	vlan-123	SnV	Rescue	Web
42:CD:BB:C0:00:00	2222::65:1	vlan-122	SnV	Evolution_X	Database
43:CD:BB:C0:00:00	10.193.102.1	vlan-124	SnV	Rescue	Database
42:CD:BB:C0:00:00	2222::65:1	vlan-127	SnV	Power_Up	Web
42:CD:BB:C0:00:00	10.193.101.1	vlan-123	SnV	Rescue	Web
43:CD:BB:C0:00:00	10.193.102.1	vlan-122	SnV	Evolution_X	Database
42:CD:BB:C0:00:00	10.193.101.1	vlan-128	SnV	Power_Up	Database
44:CD:BB:C0:00:00	2222::66:2	vlan-125	SnV	Chaos	Web
44:CD:BB:C0:00:00	2222::66:2	vlan-127	SnV	Power_Up	Web
44:CD:BB:C0:00:00	10.193.102.2	vlan-122	SnV	Evolution_X	Database
44:CD:BB:C0:00:00	2222::66:2	vlan-123	SnV	Rescue	Web
44:CD:BB:C0:00:00	2222::66:2	vlan-128	SnV	Power_Up	Database
44:CD:BB:C0:00:00	10.193.102.2	vlan-124	SnV	Rescue	Database
44:CD:BB:C0:00:00	2222::66:2	vlan-126	SnV	Chaos	Database
44:CD:BB:C0:00:00	10.193.102.2	vlan-121	SnV	Evolution_X	Web

Figure 9-8 *Output of the Python Script from Example 9-7*

Cisco UCS Manager

Cisco *Unified Computing System (UCS)* is a generic term that encompasses most of the Cisco compute products. The first UCS products were released in 2009, and they quickly established themselves as leaders in the data center compute and server market. Cisco UCS provides a unified server solution that brings together compute, storage, and networking into one system. Initially, the UCS solution took advantage of network-attached storage (NAS) or storage area networks (SANs) to support requirements for large datastores. However, with the release of Cisco HyperFlex and the hyperconverged servers, large storage datastores are included with the UCS solution.

Cisco UCS B-series blade servers, C-series rack servers, S-series storage servers, UCS Mini, and Cisco HyperFlex hyperconverged servers are all managed through one interface. This interface is provided by UCS Manager. UCS Manager provides unified, embedded management of all software and hardware components of the Cisco Unified Computing System. Cisco UCS Manager software runs on a pair of hardware appliances called *fabric interconnects*. The two fabric interconnects form an active-standby cluster that provides high availability. The UCS infrastructure that is managed by UCS Manager forms a UCS fabric that can include up to 160 servers. The system can scale to thousands of servers by integrating individual UCS Managers with Cisco UCS Central in a multidomain Cisco UCS environment.

UCS Manager participates in the complete server lifecycle, from server provisioning, device discovery, inventory, configuration, diagnostics, monitoring, fault detection, auditing, and statistics collection. All infrastructure that is managed by UCS Manager is either directly connected to the fabric interconnects or through fabric extenders. Fabric extenders, as the name implies, have the function of offering additional scalability in connecting servers back to the fabric interconnects. These zero-management, low-cost, and low-power devices eliminate the need for expensive top-of-rack Ethernet and Fibre Channel switches. Figure 9-9 illustrates how all these components connect to each other.

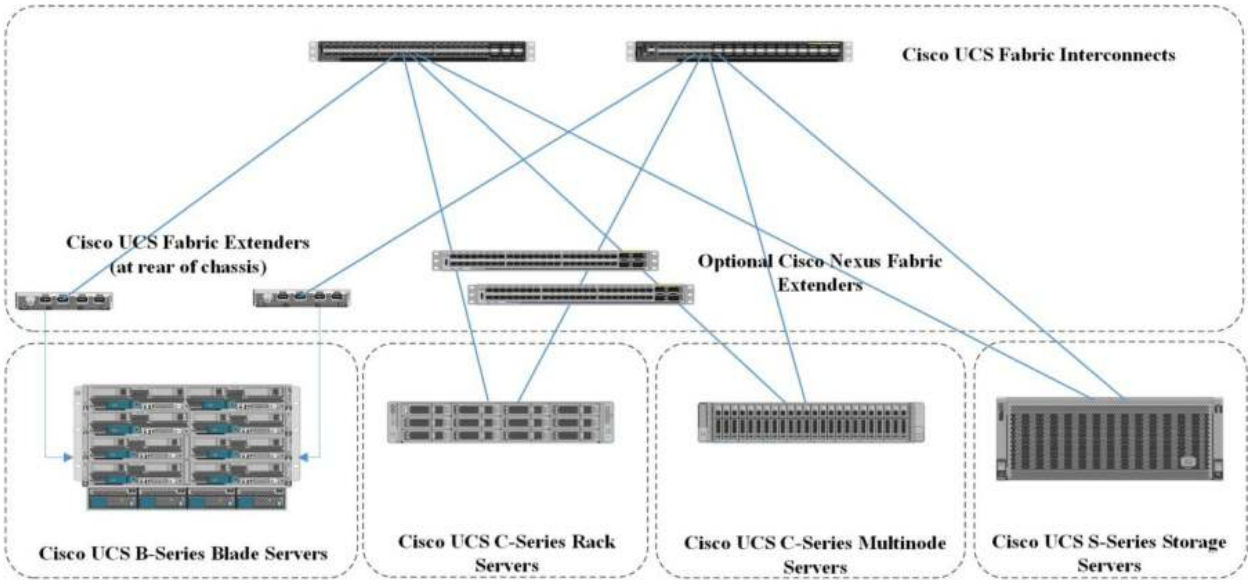


Figure 9-9 *Cisco Unified Computing System Connectivity*

All Cisco UCS servers support Cisco SingleConnect technology. Cisco SingleConnect is a revolutionary technology that supports all traffic from the servers (LAN, SAN, management, and so on) over a single physical link. The savings that this technology brings only as part of the cabling simplification is orders of magnitude higher than competing products.

Cisco UCS Manager provides an HTML-5 GUI, a CLI, and a comprehensive API. All Cisco UCS fabric functions and managed objects are available over the UCS API. Developers can take advantage of the extensive API and can enhance the UCS platform according to their unique requirements. Tools and software integrations with solutions from third-party vendors like VMware, Microsoft, and Splunk are already publicly available. Later in this chapter, we will briefly see how the Cisco UCS PowerTool for UCS Manager and the Python SDK can be used to automate and programmatically manage Cisco UCS Manager.

With Cisco UCS Manager, the data center servers can be managed within an infrastructure-as-code (IaC) framework. This is possible through another innovation that is included with the Cisco UCS solution: the service profile. The service profile is a logical construct within UCS Manager that contains the complete configuration of a physical server. All the elements of a server configuration—including RAID levels, BIOS settings, firmware revisions and settings, adapter settings, network and storage settings, and data center connectivity—are included in the service profile. When a service profile is associated with a server, Cisco UCS Manager automatically configures the server, adapters, fabric extenders, and fabric interconnects to match the configuration specified in the service profile. With service profiles, infrastructure can be provisioned in minutes instead of days. With service profiles, you can even pre-provision servers and have their configuration ready before they are even connected to the network. Once the servers come online and get discovered by UCS Manager, the service profile can be automatically deployed to the server.

The UCS Manager programmatic interface is the XML API. The Cisco UCS Manager XML API accepts XML documents over HTTP or HTTPS connections. Similar to Cisco ACI, the configuration and state information for Cisco UCS is stored in a hierarchical tree structure known as the management information tree (MIT). The MIT containing all the managed objects in the Cisco UCS system is accessible through the XML API. Any programming language can be used to generate XML documents that contain the API methods. One or more managed objects can be changed with one API call. When multiple objects are configured, the API operation stops if any of the MOs cannot be configured, and the system rolls back fully to the state of the system before the change was initiated. API operations are transactional and are done on the single data model that represents the whole system. Cisco UCS is responsible for all endpoint communications, making UCS Manager the single source of truth. Users cannot communicate directly with the endpoints, relieving developers from administering isolated, individual component configurations. All XML requests are asynchronous and terminate on the active Cisco UCS Manager.

All the physical and logical components that make up Cisco UCS are represented in a hierarchical management information tree, also known as a Management Information Model (MIM). Each node in the tree represents a managed object (MO) or a group of objects that contains its administrative and operational states. At the top of the hierarchical structure is the sys object that contains all the parent and child nodes in the tree. Each object in Cisco UCS has a unique distinguished name that describes the object and its place in the tree. The information model is centrally stored and managed by a process running on the fabric interconnects and called a data management engine (DME). When an administrative change is initiated to a Cisco UCS component, the DME first applies that change to the information model and then applies the change to the actual managed endpoint. As was the case with Cisco ACI, this approach is called a *model-driven framework*.

A specific managed object in the MIT can be identified by its distinguished name (DN) or by its relative name (RN). The DN specifies the exact managed object on which the API call is operating and consists of a series of relative names:

```
DN = {RN}/{RN}/{RN}/{RN}…
```

The relative name identifies an object within the context of its parent object.

The Cisco UCS Manager XML API model includes the following programmatic entities:

- **Classes:** Classes define the properties and states of objects in the MIT.
- **Methods:** Methods define the actions that the API performs on one or more objects.
- **Types:** Types are object properties that map values to the object state.

Several types of methods are available with the XML API:

- **Authentication methods:** These methods are used to authenticate and maintain a session:
  - **aaaLogin:** Operates as the login method
  - **aaaRefresh:** Refreshes the authentication cookie
  - **aaaLogout:** Exits the session and deactivates the corresponding authentication cookie
- **Query methods:** These methods are used to obtain information on the current configuration state of an object:
  - **configResolveDn:** Retrieves objects by DN
  - **configResolveClass:** Retrieved objects of a given class
  - **configResolveParent:** Retrieves the parent object of an object
- **Configuration methods:** These methods are used to make configuration changes to managed objects:
  - **configConfMo:** The configuration affects a single MO.
  - **configConfMos:** The configuration affects multiple subtrees.

Since the query methods available with the XML API can return large sets of data, supported filters limit this output only to subsets of information. Four types of filters are available:

- **Simple filters:** The true or false filters limit the result set of objects with the Boolean value of true or false.
- **Property filters:** These filters use the values of an object's properties as the inclusion criteria in a result set (for example, and equal filter, not equal filter, greater than filter)
- **Composite filters:** These filters are composed of two or more component filters (for example, AND filter, OR filter)
- **Modifier filter:** This filter changes the results of a contained filter. Currently, only the NOT filter is supported. This filter negates the result of a contained filter.

External applications can get Cisco UCS Manager state change information either by regular polling or by event subscription. Full event subscription is supported with the API and is the preferred method of notification. Polling usually consumes a lot of resources and should be used only in limited situations.

Cisco UCS Manager provides a managed object browser called Visore. You can access Visore by navigating to https://<UCS-Manager-IP>/visore.html. The web interface looks like the example shown in Figure 9-10.

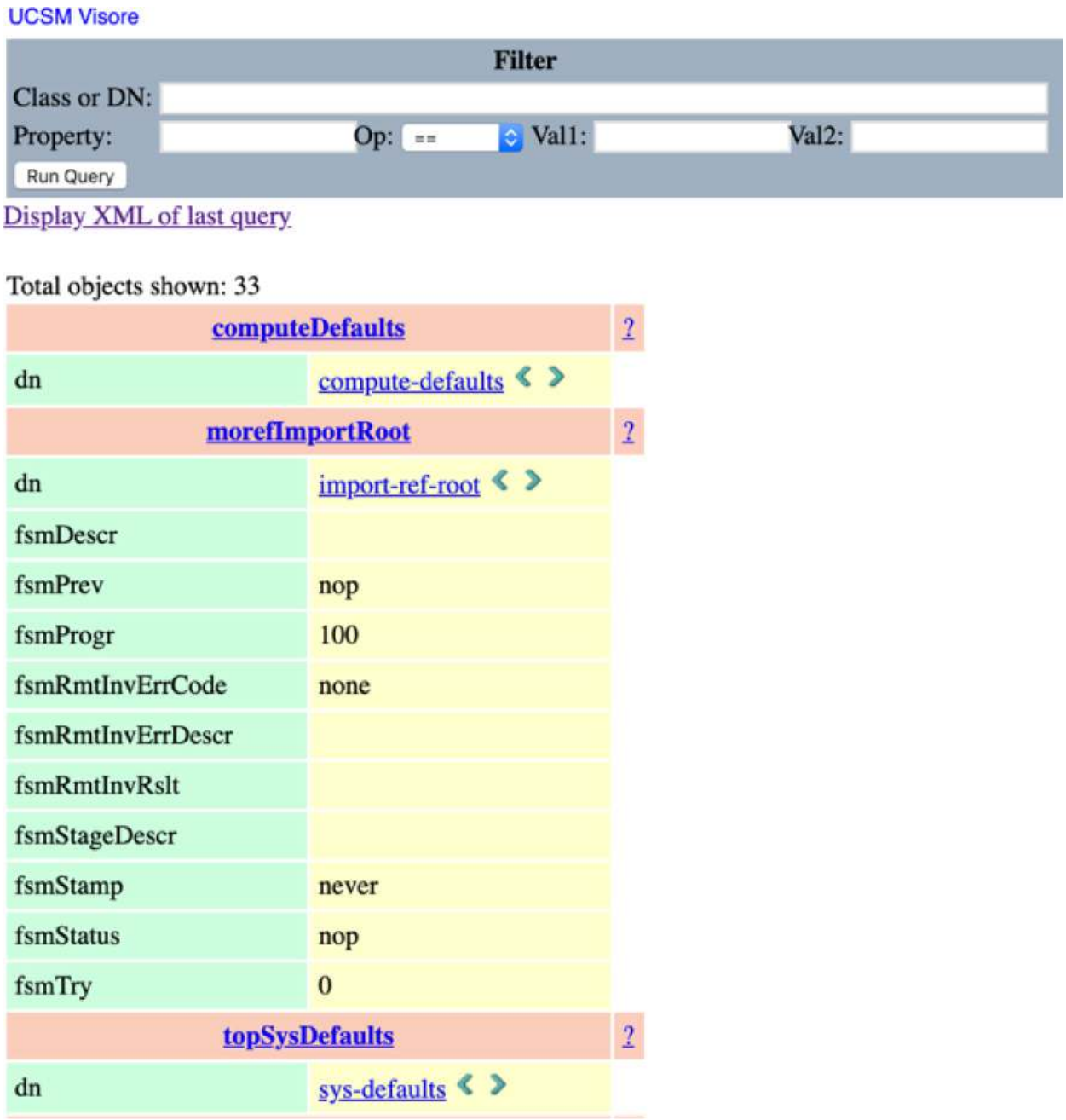


Figure 9-10 *Cisco UCS Manager Visore Interface*

You can explore the whole MIT and also can run queries for specific DNs from this interface. For additional developer resources regarding Cisco UCS Manager, see the Cisco DevNet website at https://developer.cisco.com/site/ucs-dev-center/.

Next, let's explore the Cisco UCS Manager XML API. You can find the complete documentation of the Cisco UCS Manager information model for different releases at https://developer.cisco.com/site/ucs-mim-ref-api-picker/. There, you can find all the managed objects, all methods, all types, all fault and FSM rules, and extensive documentation for each of them.

For data center administrators and developers to become more familiar with the Cisco UCS system, Cisco has released a software emulator. Cisco UCS Platform Emulator is the Cisco UCS Manager application bundled into a virtual machine. The VM includes software that emulates hardware communications for the Cisco UCS system. The Cisco UCS Platform Emulator can be used to create and test a supported Cisco UCS configuration or to duplicate an existing Cisco UCS environment for troubleshooting and development purposes. The Cisco UCS Platform Emulator is delivered as an .ova file and can run in most virtual environments. The complete Cisco UCS Manager information model documentation is also bundled within the UCS Platform Emulator. You can find the latest version of the Cisco UCS Platform Emulator at https://community.cisco.com.

In the following examples, a UCS Platform Emulator was installed and configured with an IP address of 10.10.20.110.

First, let's authenticate and get an authentication cookie. You can use **curl** to accomplish this, as follows:

```
curl -k -X POST https://10.10.20.110/nuova \

  -H 'Content-Type: application/xml' \

  -d '<aaaLogin inName='ucspe' inPassword='ucspe'></aaaLogin>'
```

The IP address of the Cisco UCS Manager is 10.10.20.110, the XML API resource is /nuova, and the authentication method that is used is aaaLogin. The username and password are passed in an XML document within the inName and inPassword variables. In this case, both the username and password are ucspe. The Content-Type header specifies the type of data that the POST call will send to the XML API, and that is XML in this case, of course.

The response should look similar to the following:

```
<aaaLogin cookie="" response="yes" outCookie="1573019916/7c901636-c461-487e-bbd0-

  c74cd68c27be" outRefreshPeriod="600" outPriv="aaa,admin,ext-lan-config,ext-lan-policy,ext-

  lan-qos,ext-lan-security,ext-san-config,ext-san-policy,ext-san-security,fault,operations,pod-

  config,pod-policy,pod-qos,pod-security,read-only" outDomains="org-root"

  outChannel="noencssl" outEvtChannel="noencssl" outSessionId="" outVersion="3.2(2.5)"

  outName="" />
```

aaaLogin specifies the method used to log in, the "yes" value confirms that this is a response, outCookie provides the session cookie, outRefreshPeriod specifies the recommended cookie refresh period (the default is 600 seconds) and the outPriv value specifies the privilege level associated with the account.

Next, let's get a list of all the objects that are part of the compute class and are being managed by this instance of Cisco UCS Manager. To accomplish this, you can use the configFindDnsByClassId method. This method finds distinguished names and returns them sorted by class ID. The **curl** command should look similar to the following:

```
curl -k -X POST https://10.10.20.110/nuova \

  -H 'Content-Type: application/xml' \

  -d '<configFindDnsByClassId

    classId="computeItem"

    cookie="1573019916/7c901636-c461-487e-bbd0-c74cd68c27be" />'
```

The XML API endpoint (https://10.10.20.110/nuova) and the Content-Type header (application/xml) stay the same. The XML data that is sent to the Cisco UCS Manager server is different. First, the configFindDnsByClassId method is specified, and then the two mandatory variables for classId and the cookie are passed in. The classId specifies the object class (in this case, it is the computeItem class), and the cookie is populated with the value of the authentication cookie obtained previously.

In this case, the response looks like the response shown in Example 9-8; it contains a complete list of all the compute items managed by the 10.10.20.110 instance of Cisco UCS Manager.

Example 9-8 *List of Compute Items That Are Managed by Cisco UCS Manager*

```
<configFindDnsByClassId cookie="1573019916/7c901636-c461-487e-bbd0-c74cd68c27be" response="yes"
classId="computeItem">
          <outDns>
                    <dn value="sys/chassis-4/blade-8"/>
                    <dn value="sys/chassis-5/blade-8"/>
                    <dn value="sys/chassis-6/blade-8"/>
                    <dn value="sys/chassis-6/blade-1"/>
                    <dn value="sys/chassis-3/blade-1"/>
                    … omitted output
                    <dn value="sys/rack-unit-9"/>
                    <dn value="sys/rack-unit-8"/>
                    <dn value="sys/rack-unit-7"/>
                    <dn value="sys/rack-unit-6"/>
                    <dn value="sys/rack-unit-5"/>
                    <dn value="sys/rack-unit-4"/>
                    <dn value="sys/rack-unit-3"/>
                    <dn value="sys/rack-unit-2"/>
                    <dn value="sys/rack-unit-1"/>
          </outDns>
</configFindDnsByClassId>
```

In this exploration of the Cisco UCS Manager XML API, next, let's get more information about a specific compute object. The method that will retrieve a single managed object for a specified DN is configResolveDn. The **curl** command for this API request should look like the one shown in Example 9-9.

Example 9-9 *Using a **curl** Command to Retrieve Information About a Compute Object*

```
curl -k -X POST https://10.10.20.110/nuova \
  -H 'Content-Type: application/xml' \
  -d '<configResolveDn
    cookie="1573019916/7c901636-c461-487e-bbd0-c74cd68c27be"
    dn="sys/chassis-4/blade-8" />'
```

Similar to the previous call, the API endpoint and Content-Type header stay the same. The XML data that is sent with the request contains the method (configResolveDn), the authentication cookie, and the DN for which additional information is requested (in this case, it is the blade number 8 in chassis number 4: sys/chassis-4/blade-8).

The response shown in Example 9-10 contains extensive information about the blade server in slot 8 from the chassis with number 4.

Example 9-10 ***curl** Command Response Containing Information About a Compute Object*

```
<configResolveDn dn="sys/chassis-4/blade-8" cookie="1573019916/7c901636-c461-487e-bbd0-c74cd68c27be"
response="yes">
.    <outConfig>
          <computeBlade adminPower="policy" adminState="in-service" assetTag="" assignedToDn=""
association="none" availability="available" availableMemory="49152" chassisId="4"
checkPoint="discovered" connPath="A,B" connStatus="A,B" descr="" discovery="complete"
discoveryStatus="" dn="sys/chassis-4/blade-8" fltAggr="0" fsmDescr="" fsmFlags=""
fsmPrev="DiscoverSuccess" fsmProgr="100" fsmRmtInvErrCode="none" fsmRmtInvErrDescr="" fsmRmtInvRslt=""
fsmStageDescr="" fsmStamp="2019-11-06T04:02:03.896" fsmStatus="nop" fsmTry="0" intId="64508"
kmipFault="no" kmipFaultDescription="" lc="undiscovered" lcTs="1970-01-01T00:00:00.000" localId=""
lowVoltageMemory="not-applicable" managingInst="A" memorySpeed="not-applicable" mfgTime="not-
applicable" model="UCSB-B200-M4" name="" numOf40GAdaptorsWithOldFw="0"
numOf40GAdaptorsWithUnknownFw="0" numOfAdaptors="1" numOfCores="8" numOfCoresEnabled="8" numOfCpus="2"
numOfEthHostIfs="0" numOfFcHostIfs="0" numOfThreads="16" operPower="off" operPwrTransSrc="unknown"
operQualifier="" operSolutionStackType="none" operState="unassociated" operability="operable"
originalUuid="1b4e28ba-2fa1-11d2-0408-b9a761bde3fb" partNumber="" policyLevel="0" policyOwner="local"
presence="equipped" revision="0" scaledMode="none" serial="SRV137" serverId="4/8" slotId="8"
totalMemory="49152" usrLbl="" uuid="1b4e28ba-2fa1-11d2-0408-b9a761bde3fb" vendor="Cisco Systems Inc"
vid=""/>
     </outConfig>
</configResolveDn>
```

While you are able to interact with the Cisco UCS Manager XML API this way, you can clearly see that doing so becomes cumbersome quickly. The preferred way of working with the XML API is either through Cisco UCS PowerTool Suite or Cisco UCS Python SDK.

The Cisco UCS PowerTool Suite is a PowerShell module that helps automate all aspects of Cisco UCS Manager. The PowerTool cmdlets work on the Cisco UCS Management Information Tree. You can use the cmdlets to execute read, create, modify, and delete operations on all the managed objects in the MIT. The Cisco UCS PowerTool suite enables easy integration with existing IT management processes and tools. You can download the PowerTool suite from https://www.cisco.com.

Cisco UCS Python SDK is a Python module that helps automate all aspects of Cisco UCS management, including server, network, storage, and hypervisor management. The Cisco UCS Python SDK works on the Cisco UCS Manager MIT, performing create, read, modify, or delete actions on the managed objects in the tree. Python versions 2.7 and higher and versions 3.5 and higher are supported. The Cisco UCS Python module for UCS Manager is called ucsmsdk. You can install it by issuing the **pip install ucsmsdk** command at the command prompt. As of this writing, the current version of the ucsmsdk module is 0.9.21.

The Cisco UCS Python SDK provides a utility called convert_to_ucs_python that gives administrators and developers the option of recording all the interactions with the Cisco UCS Manager GUI and saving them into an XML file. Running this XML file through the convert_to_ucs_python tool automatically generates the Python code corresponding to the actions that were performed in the GUI. Through this process, data center automation efforts can be sped up orders of magnitude, and simple tasks like creating a new VLAN or complex tasks like configuring a service policy template can be automated within seconds.

Next, let's explore the Cisco UCS Python SDK and see how to connect to a Cisco UCS Manager instance, retrieve a list of all the compute blades in the system, and extract specific information from the returned data. The sample Python code is built in Python 3.9.6 using version 0.9.21 of the ucsmsdk module.

First, the UcsHandle class is imported. An instance of this class is used to connect to Cisco UCS Manager. The Cisco UCS Manager IP address, username, and password are passed in as parameters to the instance of the UcsHandle class that is called handle. Several methods are available with the UcsHandle class. In this script, only three are used:

- handle.login() is used to log in to Cisco UCS Manager.
- handle.query_classid() is used to query the MIT for objects with a specific class ID.
- handle.logout() is used to log out from the Cisco UCS Manager.

The blades variable contains a dictionary of all the compute blades that are managed by the 10.10.20.110 instance of Cisco UCS Manager. Within a recursive for loop, specific information regarding the DN, serial number, administrative state, model number, and total amount of memory for each blade is extracted and displayed to the console. The Python script using the Cisco UCS Manager SDK that accomplishes all of these tasks looks like the script shown in Example 9-11.

Example 9-11 *ucsmsdk Python Example*

```
#! /usr/bin/env python
from ucsmsdk.ucshandle import UcsHandle
handle = UcsHandle("10.10.20.110","ucspe","ucspe")

# Login to Cisco UCS Manager
handle.login()

# Retrieve all the compute blades
blades = handle.query_classid("ComputeBlade")

print('{0:23s}{1:8s}{2:12s}{3:14s}{4:6s}'.format("DN",\
    "SERIAL", "ADMIN STATE", "MODEL", "TOTAL MEMORY"))
print('-'*70)

# Extract DN, serial number, admin state, model and total memory for each blade
for blade in blades:
    print('{0:23s}{1:8s}{2:12s}{3:14s}{4:6s}'.format(blade.dn,\
        blade.serial, blade.admin_state, blade.model, blade.total_memory))

# Logout from Cisco UCS Manager
handle.logout()
```

The results of running this script look like the example shown in Figure 9-11.

```
DN                      SERIAL  ADMIN STATE MODEL         TOTAL MEMORY
----------------------------------------------------------------------
sys/chassis-3/blade-1   SRV126  in-service  UCSB-EX-M4-1  49152
sys/chassis-3/blade-3   SRV127  in-service  UCSB-EX-M4-1  49152
sys/chassis-3/blade-5   SRV128  in-service  UCSB-EX-M4-1  49152
sys/chassis-3/blade-7   SRV129  in-service  UCSB-EX-M4-1  49152
sys/chassis-4/blade-1   SRV130  in-service  UCSB-B200-M4  49152
sys/chassis-4/blade-2   SRV131  in-service  UCSB-B200-M4  49152
sys/chassis-4/blade-3   SRV132  in-service  UCSB-B200-M4  49152
sys/chassis-4/blade-4   SRV133  in-service  UCSB-B200-M4  49152
sys/chassis-4/blade-5   SRV134  in-service  UCSB-B200-M4  49152
sys/chassis-4/blade-6   SRV135  in-service  UCSB-B200-M4  49152
sys/chassis-4/blade-7   SRV136  in-service  UCSB-B200-M4  49152
sys/chassis-4/blade-8   SRV137  in-service  UCSB-B200-M4  49152
sys/chassis-5/blade-1   SRV138  in-service  UCSB-B200-M5  49152
sys/chassis-5/blade-2   SRV139  in-service  UCSB-B200-M5  49152
sys/chassis-5/blade-3   SRV140  in-service  UCSB-B200-M5  49152
sys/chassis-5/blade-4   SRV141  in-service  UCSB-B200-M5  49152
sys/chassis-5/blade-5   SRV142  in-service  UCSB-B200-M5  49152
sys/chassis-5/blade-6   SRV143  in-service  UCSB-B200-M5  49152
sys/chassis-5/blade-7   SRV144  in-service  UCSB-B200-M5  49152
sys/chassis-5/blade-8   SRV145  in-service  UCSB-B200-M5  49152
sys/chassis-6/blade-1   SRV146  in-service  UCSB-B200-M4  49152
sys/chassis-6/blade-2   SRV147  in-service  UCSB-B200-M4  49152
sys/chassis-6/blade-3   SRV148  in-service  UCSB-B200-M4  49152
sys/chassis-6/blade-4   SRV149  in-service  UCSB-B200-M4  49152
sys/chassis-6/blade-5   SRV150  in-service  UCSB-B200-M4  49152
sys/chassis-6/blade-6   SRV151  in-service  UCSB-B200-M4  49152
```

Figure 9-11 *Output of the Python Script from Example 9-11*

Cisco Intersight

The Cisco Intersight platform provides intelligent cloud-powered infrastructure management for Cisco UCS and Cisco HyperFlex platforms. Cisco UCS and Cisco HyperFlex use model-based management to provision servers and the associated storage and networking automatically. Cisco Intersight works with Cisco UCS Manager and Cisco Integrated Management Controller (IMC) to bring the model-based management of Cisco compute solutions into one unified management solution. Cisco Intersight offers flexible deployment options either as Software as a Service (SaaS) on https://intersight.com or running a Cisco Intersight virtual appliance on premises. Some of the benefits of using Cisco Intersight are as follows:

- It simplifies Cisco UCS and Cisco HyperFlex management with a single management platform.
- You are able to scale across data center and remote locations without additional complexity.
- You can automate the generation and forwarding of technical support files to the Cisco Technical Assistance Center to accelerate the troubleshooting process.
- Full programmability and automation capabilities are available through a REST API interface.
- A streamlined upgrade process is available for Cisco UCS and Cisco HyperFlex.

To get started with Cisco Intersight, follow these steps:

1. Log in to https://intersight.com using a Cisco ID account.
2. After you're logged in, claim a device for the account. Endpoint devices connect to the Cisco Intersight portal through a device connector that is embedded in the management controller of each system.
3. (Optional) Add additional users to the new account. Read-only and admin roles are supported.

Cisco Intersight includes a REST API interface built on top of the OpenAPI specification. You can find the API documentation, API schemas, and SDKs at https://intersight.com/apidocs. At this writing, Python and PowerShell SDKs are available for download at the previous link. The API accepts and returns messages that are encapsulated in JSON documents and are sent over HTTPS. The Intersight API is a programmatic interface to the Management Information Model similar to Cisco ACI and Cisco UCS Manager. Just like Cisco ACI and Cisco UCS Manager, the Cisco Intersight Management Information Model is composed of managed objects. Managed objects or REST API resources are uniquely identified by uniform resource identifier (URI) or, as seen earlier in this chapter, distinguished name (DN). Managed object examples include Cisco UCS servers; Cisco UCS fabric interconnects; Cisco HyperFlex nodes and clusters; VLANs and VSANs; server, network, and storage policies; alarms; statistics; users; and roles. Cisco Intersight managed objects are represented using a class hierarchy specified in the OpenAPI specification. All the API resources are descendants of the mo.Mo class. Table 9-2 shows the properties that are common to all managed objects.

Table 9-2 *Managed Objects Properties*

Property Name	Description
Moid	A unique identifier of the managed object instance.
ObjectType	The fully qualified class name of the managed object.
AccountMoid	The Intersight account ID for the managed object.
CreateTime	The time when the managed object was created.
ModTime	The time when the managed object was last modified. The ModTime is automatically updated whenever at least one property of the managed object is modified.
Owners	An array of owners, which represents effective ownership of the object
Tags	An array of tags, which allows you to add key/value metadata to managed objects.
Ancestors	An array containing the MO references of the ancestors in the object containment hierarchy.
Parent	The direct ancestor of the managed object in the containment hierarchy.

Every managed object has a unique Moid identifier assigned when the resource is created. The Moid is used to uniquely distinguish a Cisco Intersight resource from all other resources. The Moid is a 12-byte string set when the resource is created.

Each managed object can be addressed using a unique URI, which includes the Moid. The URI can be used in any HTTP request to address the managed object. A generic Cisco Intersight URI has the following form:

https://intersight.com/*path*[?*query*]

The URI of a managed object is made up of the following parts:

- The HTTPS protocol
- The Cisco Intersight hostname—in this case, intersight.com
- The path, organized in hierarchical form
- An optional query, mentioned after the question mark and typically used to limit the output of the response to only specific parameters

For example, the URI of an object with Moid 48601f85ae74b80001aee589 could be:

https://intersight.com/api/v1/asset/DeviceRegistrations/48601f85ae74b80001aee589

Every managed object in the Cisco Intersight Information model supports tagging. Tagging is used to categorize the objects by a certain common property, such as owner, geographic location, or environment. Tags can be set and queried through the Intersight API. Each tag consists of a key and an optional value. Both the key and the value are of type string.

Managed objects may include object relationships, which are dynamic links to REST resources. Cisco Intersight uses Hypermedia as the Engine of Application State (HATEOAS) conventions to represent object relationships. Object relationships can be links to self or links to other managed objects, which, taken as a whole, form a graph of objects. Using relationships as a first-class attribute in the object model, many classes of graphs can be represented, including trees and cyclic or bipartite graphs.

Intersight provides a rich query language based on the OData standard. The query language is represented using URL query parameters for GET results. Several types of data are supported with the Intersight queries, such as string, number, duration, data and time, and time of day.

When a client sends an API request, the Intersight web service must identify and authenticate the client. The Intersight web service supports two authentication methods:

- API keys
- Session cookies

An Intersight API key is composed of a keyId and a keySecret. The API client uses the API key to cryptographically sign each HTTP request sent to the Intersight web service. The "signature" parameter is a base 64–encoded digital signature of the message HTTP headers and message content. API keys are generated in the Settings > API section of the Intersight web interface. As a best practice, it is recommended to generate separate API keys for each client application that needs access to the API.

Cookies are used primarily by the Intersight GUI client running in a browser. When accessing the Intersight web service, end users must first authenticate to https://sso.cisco.com. When authentication is successful, sso.cisco.com sends a signed SAML assertion to the Intersight web service, and Intersight generates a session cookie with a limited time span validity. The client must send the session cookie in each API request.

Included with the Cisco Intersight REST API documentation at https://intersight.com/apidocs is the API reference documentation and an embedded REST API client. Figure 9-12 displays the web interface for the Cisco Intersight API reference documentation. In the figure, the "Get a list of 'equipmentDeviceSummary' instances" API call is selected. The query parameters that this specific API call supports are displayed; also displayed is the API URI for the endpoint that will return a list of all the devices that are managed by Cisco Intersight for this specific account. Similar to Postman, if the Send button is clicked, the API call is triggered, and the response is displayed in the Response Text window.

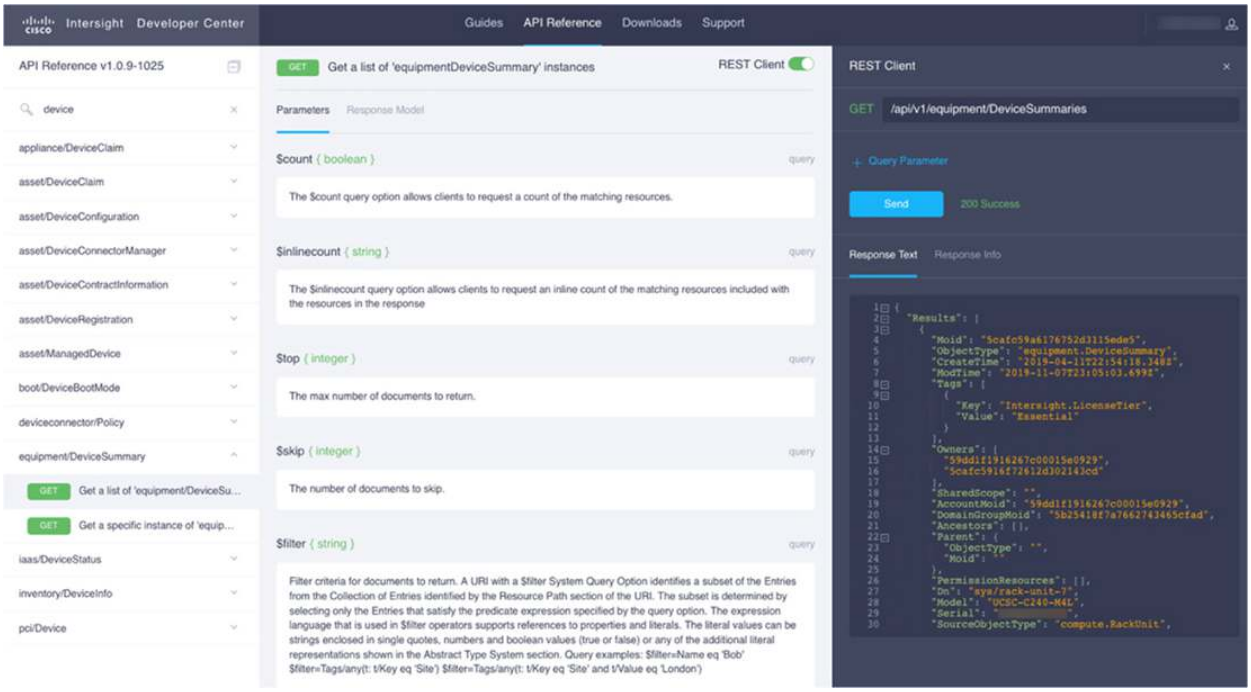


Figure 9-12 *Cisco Intersight REST API Reference Documentation*

Under the Downloads section of the https://intersight.com/apidocs, you can download the Cisco Intersight Python and PowerShell SDKs. The Python SDK covers all the functionality of the REST API and offers Python classes and methods that can be used to simplify Cisco Intersight automation projects. The Python sample code in Example 9-12 was developed using the Intersight module version 1.0 and Python 3.9.6. Trying to replicate the earlier REST API call (equipmentDeviceSummary) that returns a list of all the devices managed by Cisco Intersight for a specific account should result in Python code similar to this example.

Example 9-12 *Intersight Python Module Example*

```
#! /usr/bin/env python

from intersight.intersight_api_client import IntersightApiClient
from intersight.apis import equipment_device_summary_api

# Create an Intersight API Client instance
api_instance = IntersightApiClient(
    host="https://intersight.com/api/v1",\
    private_key="/Path_to/SecretKey.txt",\
    api_key_id="your_own_api_key_id")

# Create an equipment device handle
d_handle = equipment_device_summary_api.EquipmentDeviceSummaryApi(api_instance)
devices = d_handle.equipment_device_summaries_get().results

print('{0:35s}{1:40s}{2:13s}{3:14s}'.format("DN", "MODEL", "SERIAL",\
     "OBJECT TYPE"))
print('-'*105)

# Loop through devices and extract data
for device in devices:
    print('{0:35s}{1:40s}{2:13s}{3:14s}'.format(device.dn, device.model,\
         device.serial, device.source_object_type))
```

The first two lines of the sample code use the import keyword to bring in and make available for later consumption the IntersightApiClient Python class that will be used to create a connection to the Cisco Intersight platform and the equipment_device_summary_api file that contains Python objects useful for retrieving equipment that is being managed by Intersight. All Cisco Intersight REST API endpoints have a corresponding Python file containing classes and methods that can be used to programmatically process those REST API endpoints. Next, an instance of the IntersightApiClient class is created to establish a connection and have a hook back to the Cisco Intersight platform. The following three parameters need to be passed in to instantiate the class:

- host, which specifies the Cisco Intersight REST API base URI.
- private_key, which specifies the path to the file that contains the keySecret of the Intersight account that will be used to sign in.
- api_key_id, which contains the keyId of the same Intersight account. As mentioned previously, both the keyId and keySecret are generated in the Intersight web interface, under Settings > API keys.

Next, an instance of the EquipmentDeviceSummaryApi class is created and stored in the d_handle variable. This Python class maps into the /api/v1/equipment/DeviceSummaries REST API resource. The d_handle variable contains the handle to that REST API resource. The equipment_device_summaries_get method, which is available with the EquipmentDeviceSummaryApi class, is invoked next, and the results are stored in the devices variable. The devices variable contains a complete list of all the equipment that is managed by Cisco Intersight for the user account with the keyId and keySecret with which the initial connection was established. The for loop iterates over each device in the list and extracts the distinguished name, model, serial number, and object type and displays them to the console. The output of this Python script for a test user account looks like Figure 9-13.

```
DN                                  MODEL                                  SERIAL       OBJECT TYPE
-------------------------------------------------------------------------------------------------------
sys/rack-unit-7                     UCSC-C240-M4L                          FCH        T  compute.RackUnit
sys/rack-unit-8                     UCSC-C240-M4L                          FCH        7  compute.RackUnit
sys/rack-unit-3                     HXAF220C-M5SX                          WZP        3  compute.RackUnit
sys/rack-unit-1                     HXAF220C-M5SX                          WZP        6  compute.RackUnit
sys/rack-unit-6                     HXAF220C-M5SX                          WZP        H  compute.RackUnit
sys/rack-unit-5                     HXAF220C-M5SX                          WZP        J  compute.RackUnit
sys/rack-unit-2                     HXAF220C-M5SX                          WZP        7  compute.RackUnit
sys/rack-unit-4                     HXAF220C-M5SX                          WZP        F  compute.RackUnit
sys/rack-unit-8/adaptor-1           UCSC-PCIE-CSC-02                       FCH        E  adapter.Unit
sys/rack-unit-7/adaptor-1           UCSC-PCIE-CSC-02                       FCH        J  adapter.Unit
sys/rack-unit-5/adaptor-1           UCSC-MLOM-C40Q-03                      FCH        4  adapter.Unit
sys/rack-unit-2/adaptor-1           UCSC-MLOM-C40Q-03                      FCH        T  adapter.Unit
sys/rack-unit-6/adaptor-1           UCSC-MLOM-C40Q-03                      FCH        H  adapter.Unit
sys/rack-unit-1/adaptor-1           UCSC-MLOM-C40Q-03                      FCH        F  adapter.Unit
sys/rack-unit-4/adaptor-1           UCSC-MLOM-C40Q-03                      FCH        W  adapter.Unit
sys/rack-unit-3/adaptor-1           UCSC-MLOM-C40Q-03                      FCH        C  adapter.Unit
sys/switch-A                        UCS-FI-6248UP                          SSI        2  network.Element
sys/switch-B                        UCS-FI-6248UP                          SSI        7  network.Element
sys/rack-unit-1/adaptor-MLOM        UCSC-MLOM-CSC-02                       FCH        B  adapter.Unit
sys/rack-unit-1                     UCSC-C240-M4SX                         FCH        A  compute.RackUnit
sys/rack-unit-1/adaptor-MLOM        UCSC-MLOM-CSC-02                       FCH        J  adapter.Unit
sys/rack-unit-1                     UCSC-C240-M4SX                         FCH        2  compute.RackUnit
sys/rack-unit-1/adaptor-MLOM        UCSC-MLOM-CSC-02                       FCH        8  adapter.Unit
sys/rack-unit-1                     UCSC-C240-M4SX                         FCH        0  compute.RackUnit
sys/rack-unit-1/adaptor-MLOM        UCSC-MLOM-CSC-02                       FCH        B  adapter.Unit
sys/rack-unit-1                     UCSC-C240-M4SX                         FCH        Z  compute.RackUnit
sys/rack-unit-1/network-adapter-L   Intel(R) I350 1 Gbps Network Controller             adapter.Unit
sys/rack-unit-1/network-adapter-L   Intel(R) I350 1 Gbps Network Controller             adapter.Unit
sys/rack-unit-1/network-adapter-L   Intel(R) I350 1 Gbps Network Controller             adapter.Unit
sys/rack-unit-1/network-adapter-L   Intel(R) I350 1 Gbps Network Controller             adapter.Unit
```

Figure 9-13 *Output of the Python Script from Example 9-12*

Exam Preparation Tasks

As mentioned in the section "How to Use This Book" in the Introduction, you have a couple of choices for exam preparation: the exercises here, Chapter 19, "Final Preparation," and the exam simulation questions on the companion website.

Review All Key Topics

Review the most important topics in this chapter, noted with the Key Topic icon in the outer margin of the page. Table 9-3 lists these key topics and the page number on which each is found.

Table 9-3 *Key Topics*

Key Topic Element	Description	Page Number
Paragraph	Cisco Nexus 9000 switches	
Paragraph	Cisco ACI fabric	
Paragraph	The configuration of the ACI fabric	
Paragraph	Cisco ACI fabric physical and logical components	
Paragraph	Cisco ACI fabric policies	
Paragraph	Endpoint groups (EPGs)	
Paragraph	The APIC REST API URI	
Paragraph	Tools and libraries for Cisco ACI automation	
Paragraph	UCS Manager and the server lifecycle	
Paragraph	The UCS Manager programmatic interface	
Paragraph	The Cisco software emulator	
Paragraph	The Cisco UCS Python SDK	
Paragraph	The Cisco Intersight REST API interface	
Paragraph	Client API request and Intersight	
Paragraph	The Cisco Intersight REST API documentation	

Define Key Terms

Define the following key terms from this chapter and check your answers in the glossary:

Application Centric Infrastructure (ACI)
Application Policy Infrastructure Controller (APIC)
Management Information Model (MIM)
management information tree (MIT)
managed object (MO)
distinguished name (DN)
virtual routing and forwarding (VRF) instance
endpoint group (EPG)
Unified Computing System (UCS)

Chapter 10. Cisco Collaboration Platforms and APIs

This chapter covers the following topics:

- **Introduction to the Cisco Collaboration Portfolio:** This section introduces the collaboration portfolio by functionality and provides an overview of the product offerings.
- **Webex API:** This section introduces Webex and the rich API for managing and creating applications, integrations, and bots.
- **Cisco Finesse:** This section provides an overview of Cisco Finesse and API categories, provides sample code, and introduces gadgets.
- **Webex Meetings APIs:** This section introduces the high-level API architecture of Webex Meetings and the Meetings XML API for creating, updating, and deleting meetings.
- **Webex Devices:** This section overviews the Webex Devices portfolio, xAPI, and sample applications to turn on device presence detectors.
- **Cisco Unified Communications Manager:** This section provides an overview of Cisco Call Manager and Cisco Administrative XML (AXL), showing sample code for using the SDK.

Millions of people rely on Cisco collaboration and Webex solutions daily to collaborate with their teams, partners, and customers. These products help them work smarter, connect across boundaries, and drive new innovative ideas forward. Cisco products offer secure, flexible, seamless, and intelligent collaboration. This chapter introduces the various products and how to integrate these collaboration products via APIs. It covers the following:

- Cisco Webex
- Cisco Webex Devices
- Cisco Unified Communications Manager
- Cisco Finesse

"Do I Know This Already?" Quiz

The "Do I Know This Already?" quiz allows you to assess whether you should read this entire chapter thoroughly or jump to the "Exam Preparation Tasks" section. If you are in doubt about your answers to these questions or your own assessment of your knowledge of the topics, read the entire chapter. Table 10-1 lists the major headings in this chapter and their corresponding "Do I Know This Already?" quiz questions. You can find the answers in Appendix A, "Answers to the 'Do I Know This Already?' Quiz Questions."

Table 10-1 *"Do I Know This Already?" Section-to-Question Mapping*

Foundation Topics Section	Questions
Introduction to the Cisco Collaboration Portfolio	1
Webex API	2–4
Cisco Finesse	5, 6
Webex Meetings APIs	7
Webex Devices	8, 9
Cisco Unified Communications Manager	10

Caution

The goal of self-assessment is to gauge your mastery of the topics in this chapter. If you do not know the answer to a question or are only partially sure of the answer, you should mark that question as wrong for purposes of self-assessment. Giving yourself credit for an answer that you correctly guess skews your self-assessment results and might provide you with a false sense of security.

1. Which of the following are part of the Cisco collaboration portfolio? (Choose three.)
   - **a.** Video Calling
   - **b.** Bots
   - **c.** Remote Expert
   - **d.** Connected Mobility Experience
2. How does Webex allow you to access APIs? (Choose three.)
   - **a.** Integrations
   - **b.** Bots
   - **c.** Drones
   - **d.** Personal access tokens
3. Guest users of Webex authenticate with guest tokens, which use _____.
   - **a.** Base64
   - **b.** No token
   - **c.** JWT
   - **d.** Sessions
4. Which of the following uses webhooks in Webex?
   - **a.** Bots
   - **b.** Guests
   - **c.** Nobody
   - **d.** Integrations
5. True or false: The Finesse desktop application is entirely built using APIs.

a. False

b. True

6. To join a meeting using the Webex API, which method and endpoint should you use?

a. POST https://webexapis.com/v1/meetings/join

b. GET https://webexapis.com/v1/meetings

c. PATCH https://webexapis.com/v1/meetings/{meetingId}

d. PUT https://webexapis.com/v1/meetings/{meetingId}

7. Which of the following enables hosts/users to update the information for a scheduled meeting that they can edit?

a. SetMeeting

b. ListMeeting

c. CreateMeeting

d. Layered systems

8. Which of the following is the application programming interface (API) for collaboration endpoint software?

a. MQTT

b. TAPI

c. DevNet

d. xAPI

9. Which of the following protocol methods can xAPI be accessed on a device? (Choose all that apply.)

a. SSH

b. FTP

c. HTTP

d. Websocket

10. Which of the following provides a mechanism for inserting, retrieving, updating, and removing data from Cisco Unified Communications Manager?

a. Session Initiation Protocol

b. Administration XML

c. Skinny

d. REST API

Foundation Topics

Introduction to the Cisco Collaboration Portfolio

Cisco's collaboration portfolio is vast, but it can be logically broken down into essentially four high-level components:

- **Unified Communications Manager:** This product unifies voice, video, data, and mobile apps.
- **Unified Contact Center:** This product provides customers with personalized omnichannel experiences.
- **Cisco Webex:** This conferencing solution enables teamwork with intuitive solutions that unite people.
- **Cisco collaboration endpoints:** These devices provide better-than-being-there experiences via new devices.

Figure 10-1 depicts the Cisco collaboration portfolio and the various products in each category.

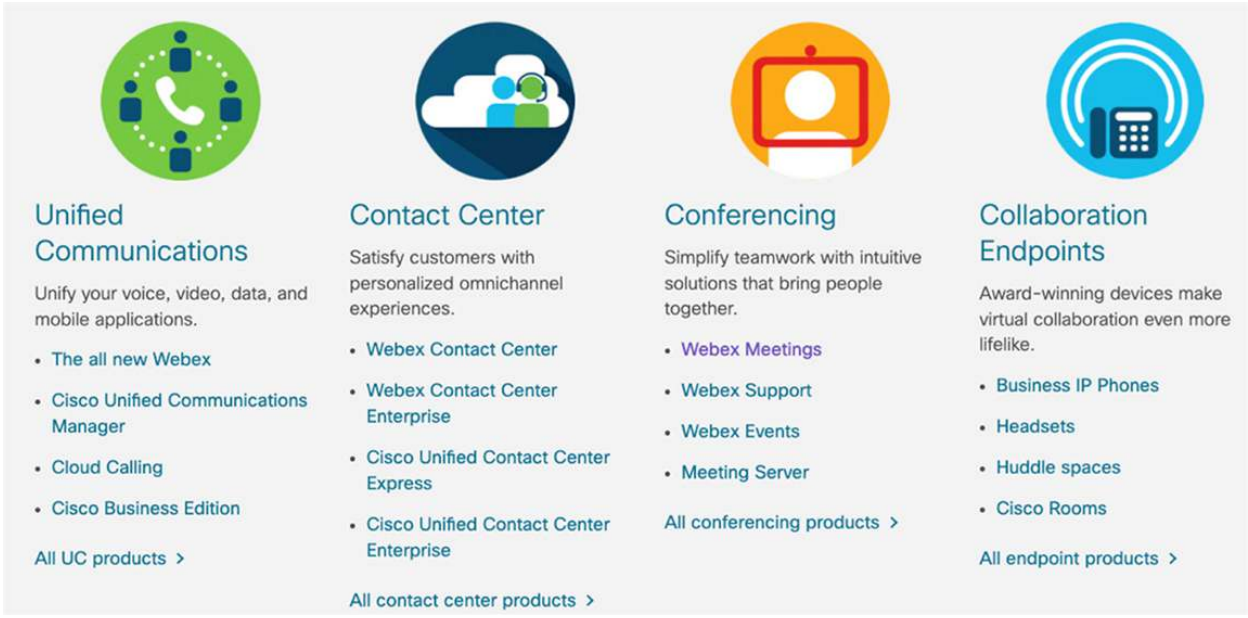


Figure 10-1 *Rich Collaboration Portfolio*

Unified Communications

People work together in different ways and use many collaboration tools, such as IP telephony for voice calling, web and video conferencing, voicemail, mobility, desktop sharing, instant messaging and presence, and more.

Cisco Unified Communications solutions deliver seamless user experiences that help people work together more effectively—anywhere and on any device. They bring real-time communication from your phone system and conferencing solutions with messaging and chat and integrate with everyday business applications using APIs.

Unified Communications solutions are available as on-premises software, partner-hosted solutions, and a service (UCaaS) from cloud providers.

The following sections describe the products available under the Unified Communications umbrella.

Cisco Webex

Cisco Webex brings people and work together in a single reimagined workspace in and beyond the meeting. Webex allows you to connect with people (inside and outside your organization) and places all your tools right in the center of your workflow. It breaks down the silos that exist for some across the collaboration experience.

Webex and Webex Meetings have a **Join** button to join a meeting quickly. This helps ensure that meetings start on time and streamlines the process of joining a meeting. Cisco Webex also has features that will help you make decisions on which meetings to prioritize and when to join them:

- **Seeing invitee status/participants:** Before joining, every invitee can see who has accepted/declined the meeting and who's in the meeting live. There is no need to switch back and forth between your calendar and the meeting application.
- **Instantly switch between meetings:** If you need to move from one session to another, simply leave the meeting with one click and join the other with another click.
- **Efficiently informing attendees when you are running late:** In Webex, you can message people in the meeting to keep them posted on your status. Gone are the days of sending an email that no one would read because they were in the meeting.

Webex provides a space for a team to start working—to discuss issues and share content before a meeting. All the discussions and work from before the meeting are available in the meeting space when the session begins. You can share content from any device in the meeting—directly from the space or another device wirelessly and quickly.

An added benefit of joining a meeting through Webex is that everyone participates equally during the session. You can mute noisy participants, record the meeting, and perform other meeting management tasks without disrupting the flow.

Cisco Webex Calling

Webex Calling is a cloud-based phone system optimized for midsize businesses and provides the essential business calling capabilities you will likely need. With Webex Calling, there's no need to worry about the expense and complexity of managing phone system infrastructure on your premises anymore. Cisco takes care of the Webex Cloud so you can focus on what matters most.

You can choose from various Cisco IP phones to make and receive calls. Enjoy the calling features you are used to from a traditional phone system to help run your organization smoothly and never miss a call. If you are a mobile worker or out of the office, you can make and receive calls on your smartphone, computer, or tablet using the Cisco Webex app.

Webex Calling seamlessly integrates with Webex and Webex Meetings, allowing you to take collaboration inside and outside your organization to a new level. Customers and business partners can make high-definition audio and video calls from the office, home, or on the go. Screen sharing, file sharing, real-time messaging, and whiteboarding can turn any call into a productive meeting. You can also add a Cisco Webex Board, Room, or Desk Device to maximize Webex and Meetings and improve teamwork.

Cisco Webex Calling delivers all the features of a traditional PBX through a monthly subscription service. Essential qualities include the following:

- An advanced set of enterprise-grade PBX features
- A rich user experience that includes the CiscoWebex Calling app for mobile and desktop users, integrated with the Cisco Webex collaboration app
- Support for an integrated user experience with Cisco Webex Meetings and Webex Devices, including Cisco IP Phones 6800, 7800, and 8800 Series desk phones and analog ATAs
- Delivery from a set of regionally distributed, geo-redundant data centers around the globe
- Service that is available across a growing list of countries in every region
- Protection of existing investment in any on-premises Cisco Unified Communications Manager (Unified CM) licenses through the Cisco Collaboration Flex Plan
- A smooth migration to the cloud at your pace through support of cloud and mixed cloud and on-premises deployments

Cisco Unified Communications Manager (Unified CM)

Cisco Unified CM, often called *Call Manager*, is the core of Cisco's collaboration portfolio. It delivers people-centric user and administrative experiences and supports a full range of collaboration services, including video, voice, instant messaging and presence, messaging, and mobility on Cisco and third-party devices. Unified CM is the industry leader in enterprise call and session management platforms, with over 300,000 customers worldwide and over 120 million Cisco IP phones and soft clients deployed.

Unified Contact Center

The Cisco Unified Contact Center (Unified CC), also called Finesse, is a next-generation desktop designed to provide the optimal user experience for agents. It is 100% browser-based, so agent client machines do not need any Unified CC[and]specific applications. Cisco Finesse is a next-generation agent and supervisor desktop designed to provide a collaborative experience for the various communities interacting with a customer service organization. It also helps improve the customer experience and offers a user-centric design to enhance customer care representative satisfaction.

Cisco Finesse provides the following:

- An agent and supervisor desktop that integrates traditional contact center functions into a thin-client desktop.
- A 100% browser-based desktop is implemented through a Web 2.0 interface; no client-side installations are required.
- A single customizable interface that gives customer care providers quick and easy access to multiple assets and information sources.
- Open Web 2.0 APIs that simplify the development and integration of value-added applications and minimize the need for detailed desktop development expertise.

Cisco Webex

Cisco Webex is a conferencing solution that allows people to collaborate more effectively with each other anytime, anywhere, and from any device. Webex online meetings genuinely engage with high-definition video. Webex makes online sessions easy and productive with features such as document, application, and desktop sharing; integrated audio/video, active speaker detection, recording, and machine learning features.

Cisco Collaboration Endpoints

To support and empower the modern workforce, Cisco has a "no-compromise" collaboration solution for every room, on every desk, in every pocket, and into every application. Its portfolio of collaboration devices includes everything from voice to collaboration room devices for small businesses to very large enterprises. Most of the portfolio has been redesigned to make collaboration more affordable, accessible, and easy to use. Many Cisco collaboration endpoints have received the Red Dot Award for excellence in design.

The Cisco collaboration endpoint portfolio includes the following devices:

- **Room-based devices** include video systems or codecs in the Cisco TelePresence MX Series, SX, and DX Series.
- **Cisco Webex Board:** Cisco Webex Board allows you to wirelessly present whiteboard and video or audio conference for team collaboration.
- **Webex Share:** The new Webex Share device allows easy, one-click wireless screen sharing from the Webex software client to any external display with an HDMI port.
- **Collaboration desktop video devices:** A range of options are available, from entry-level HD video up to the lifelike DX video collaboration devices.
- **IP Phone portfolio:** Cisco IP Phone devices deliver superior voice communications, with select endpoints supporting HD video and a range of options that offer choices for businesses of various sizes and with unique needs and budgets. The portfolio also supports specialty use cases, such as on-campus wireless with WLAN handsets and an audio-conferencing endpoint for small to large conference rooms. The IP Phone portfolio aims to deliver the highest-quality audio communications with affordable, scalable options for desktop video that are easy to use and administer so you can collaborate effectively and achieve your desired business results.
- **Cisco Headset 300/500/700/900 Series:** These headsets deliver surprisingly vibrant sound for open workspaces. Users can stay focused in noisy environments with rich sound, exceptional comfort, and proven reliability. The headsets offer a lightweight form factor designed for workers who spend much time collaborating in contact centers and open workspaces. With the USB headset adapter, the headset Series delivers an enhanced experience, including automatic software upgrades, in-call presence indicators, and audio customizations that allow you to adjust how you hear the far end and how they hear you. For the latest information on headsets, it's ways best to visit - https://www.cisco.com/c/en/us/products/collaboration-endpoints/headsets/index.html

Cisco now provides more intelligent audio, video, and usability, offering new ways for users to bring their devices, such as smartphones or tablets, into a phone or video meeting to further enhance the collaboration experience.

API Options in the Cisco Collaboration Portfolio

The Cisco collaboration portfolio is rich in features. APIs are used to integrate and scale these products when building various applications. The following sections cover four categories of collaboration APIs:

- Webex Meetings APIs
- Webex
- Contact Center (Finesse)
- Endpoints

Webex API

Webex makes it easy for everyone on a team to be in sync. Conversations in Webex take place in virtual meeting rooms called *spaces*. Some spaces live for a few hours, while others become permanent fixtures of a team's workflow. Webex allows seamless conversations between messages, video calls, and real-time whiteboarding sessions.

Getting started with the Webex APIs is easy. These APIs allow developers to build integrations and bots for Webex and allow administrators to perform administrative tasks.

Webex APIs provide applications with direct access to the Cisco Webex platform, giving you the ability to do the following:

- Administer the Webex platform for an organization, add user accounts, and so on
- Create a Webex team, space, and memberships
- Search for people in the company
- Post messages in a Webex space
- Get Webex space history or be notified in real-time when others post new messages

Figure 10-2 shows an overall picture of how Webex is organized. Only an admin can add a new organization or a new user account. An organization comprises teams, and a team can have one or more rooms. A person is an end user who can be added to a room. People communicate with each other or everyone else in a room via messages.

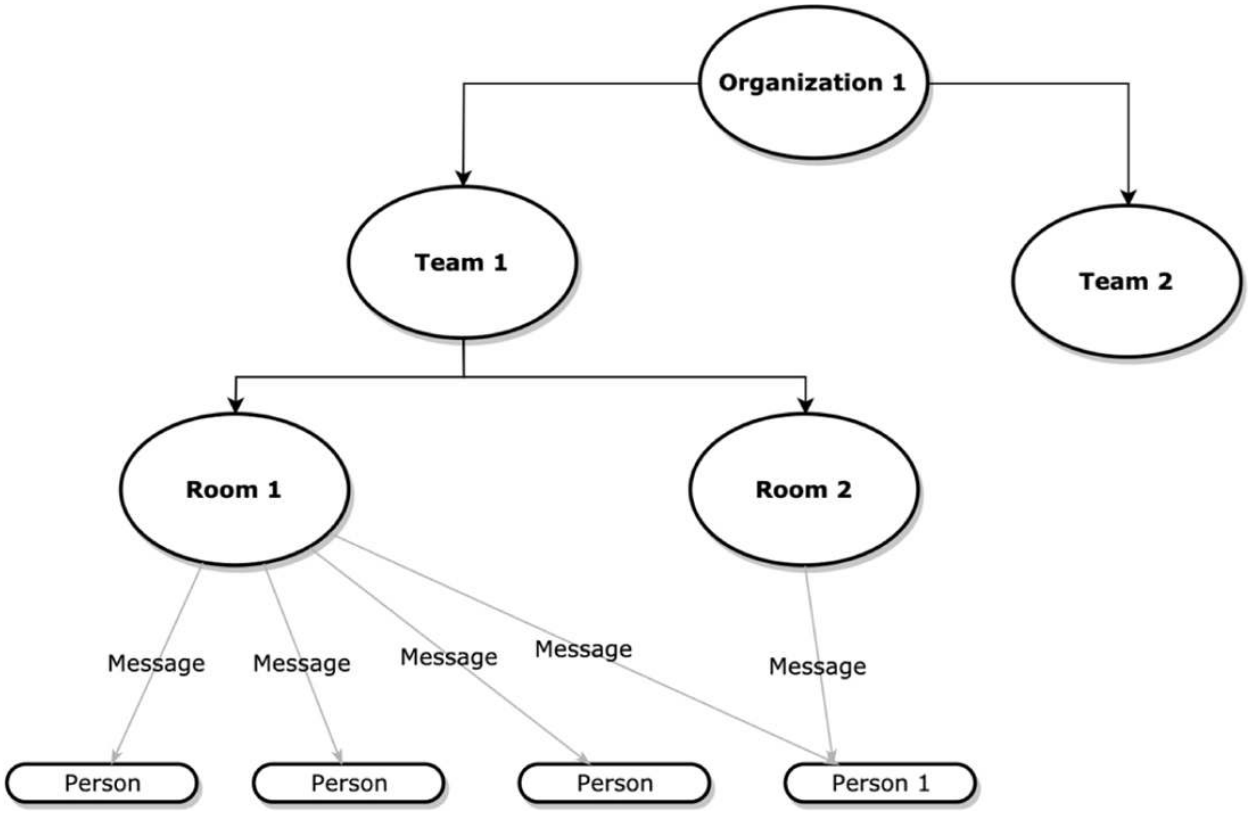


Figure 10-2 *Webex Building Blocks: Organizations, Teams, Rooms, People, and Messages*

API Authentication

There are four ways to access the Webex APIs:

- Personal access tokens
- Integrations
- Bots
- Guest issuers

The Webex Representational State Transfer (REST) API uses the Cisco Webex Common Identity (CI) account. Once you create an account to join Webex, you can access the Common Identity account, which allows you to use the APIs and SDKs.

Personal Access Tokens

When making requests to the Webex REST API, an authentication HTTP header is used to identify the requesting user. This header must include an access token, a personal access token from the developer site (https://developer.Webex.com), a bot token, or an OAuth token from integration or a guest issuer application. The API reference uses your access token, which you can use to interact with the Webex API. This token has a short lifetime—. It lasts only 12 hours after logging in to the site—, so it shouldn't be used outside of app development. Figure 10-3 shows the bearer token as obtained from the developer portal.

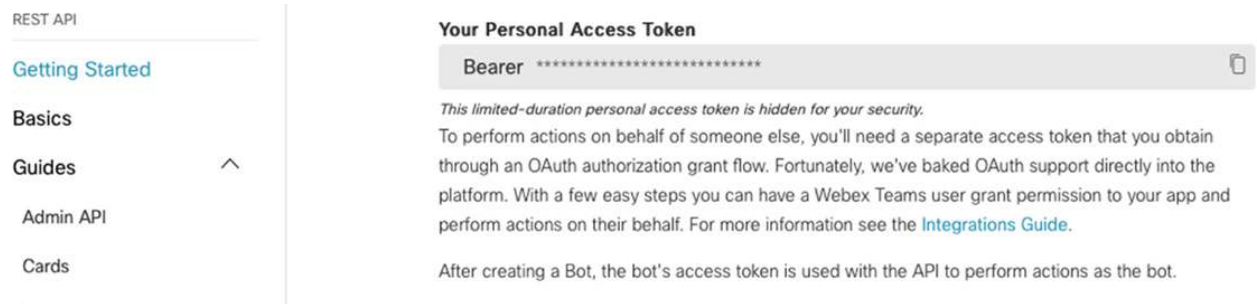


Figure 10-3 *Webex: Getting a Personal Access Token*

Integrations

To perform actions on behalf of someone else, you need a separate access token that you obtain through an OAuth authorization grant flow. OAuth is supported directly into the platform. With a few easy steps, you can have a Webex user grant permission to

your app and perform actions on that person's behalf. Figure 10-4 shows how third-party apps can access the platform.

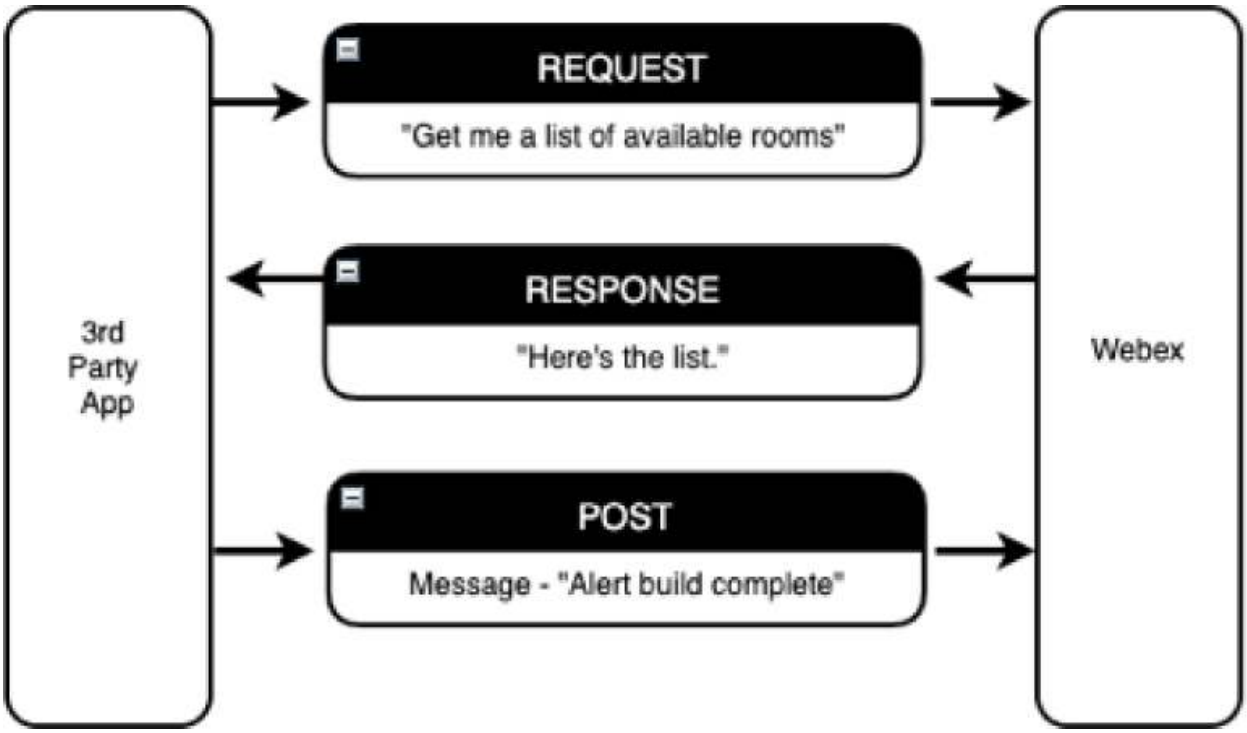


Figure 10-4 *Webex: Third-Party Integrations*

You use an integration to request permission to invoke the Webex REST API on behalf of another Webex user. To provide security, the API supports the OAuth 2 standard, which allows a third-party integration to get a temporary access token for authenticating API calls instead of asking users for their password.

Here are a few easy steps to get started using an integration:

Step 1. Register an integration with Webex at https://developer.Webex.com/my-apps/new. Figure 10-5 shows the sample form on the portal that allows you to create a new integration.

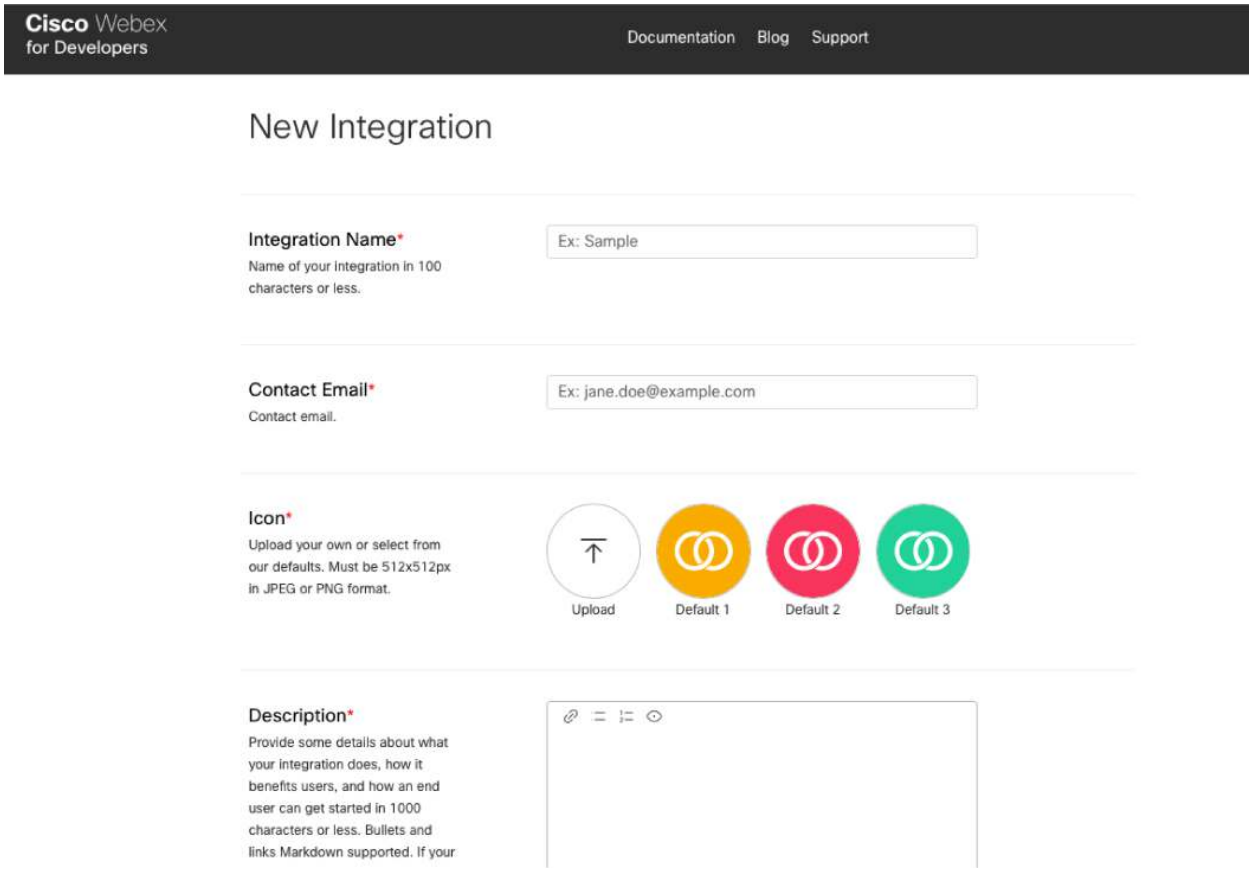


Figure 10-5 *Creating a New Integration via the Developer Portal*

Step 2. Request permission by using an OAuth grant flow by invoking the flow via https://webexapis.com/v1/authorize and providing a redirect URL to come back to. After the integration is created successfully, you see a screen like the one in Figure 10-6, which also shows the full authorization URL.

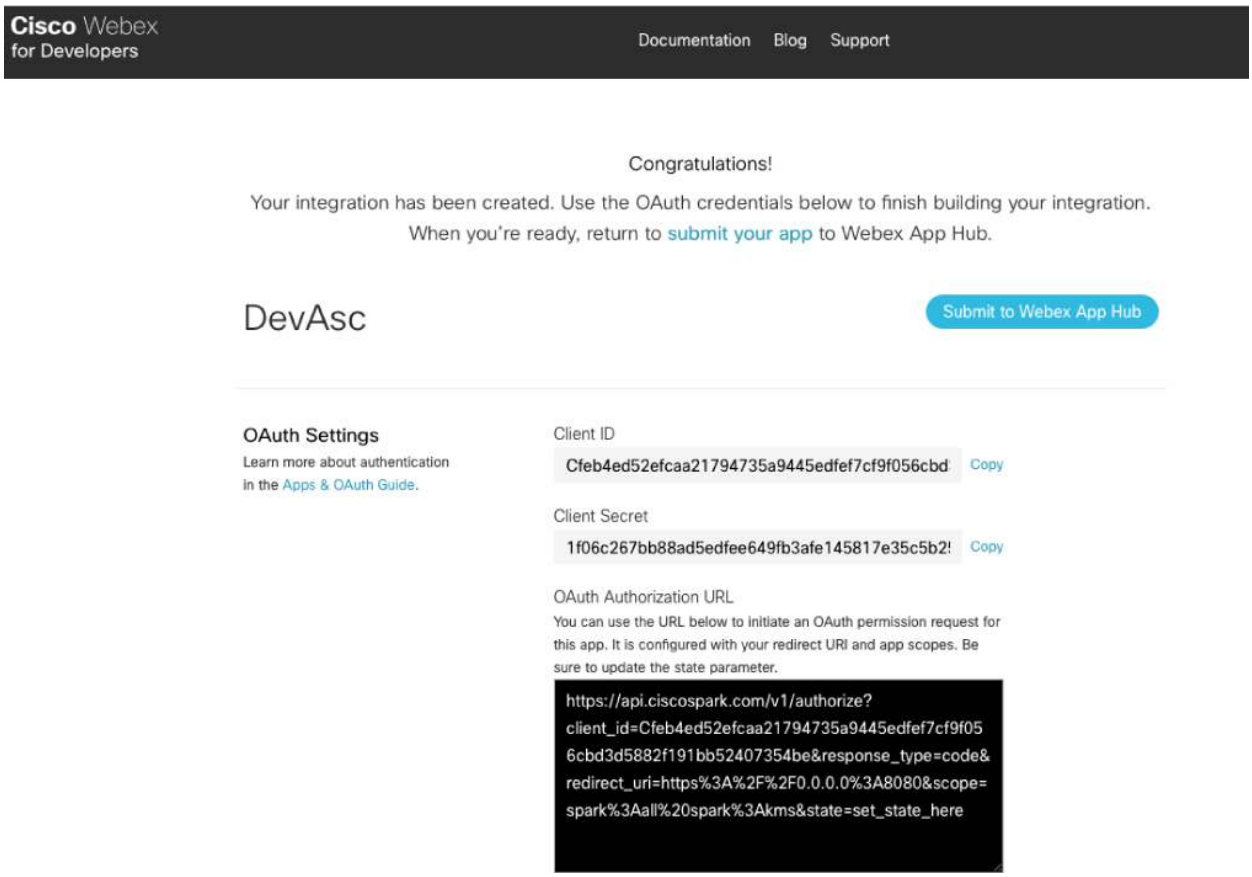


Figure 10-6 *Successful Integration Results in OAuth Credentials*

Step 3. On the Figure 10-7 screen, click **Accept** to obtain the authorization code for an access token.

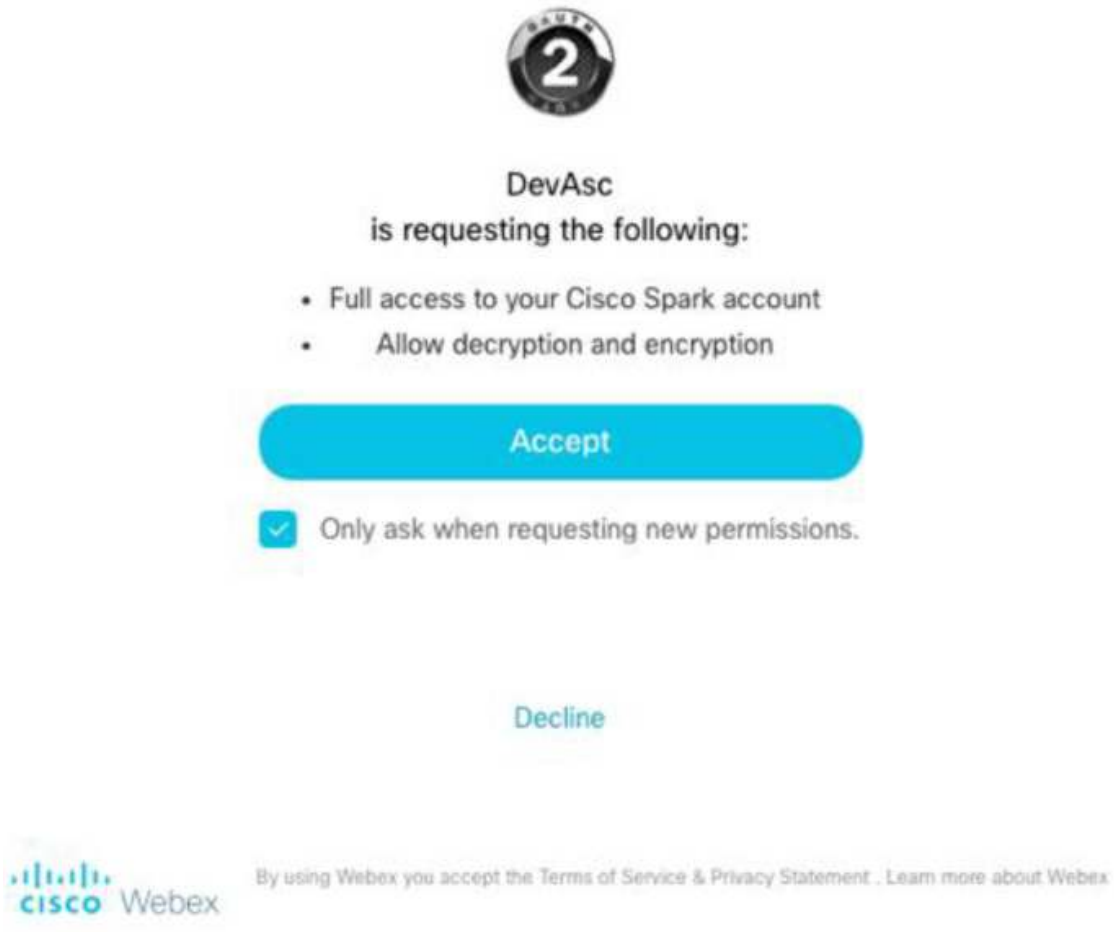


Figure 10-7 *Using the OAuth Credentials and Accepting Permissions*

The redirect URL contains a code parameter in the query string like so:

https://0.0.0.0:8080/?code=NzAwMGUyZDUtYjcxMS00YWM4LTg3ZDYtNzdhMDhhNWRjZGY5NGFmMjA3ZjEtYzRk_PF84_1eb65fdf-9643-417f-9974-ad72cae0e10f&state=set_state_here

Access Scopes

Scopes define the level of access that an integration requires. Each integration alerts the end user about its scope. Scopes also determine what resources the Access Token has access to. Table 10-2 lists and describes the scopes.

Table 10-2 *Webex Scopes API Definitions*

Scope	Description
permission dialog.spark:all	Full access to your Webex account
spark:people_read	Read your company directory
spark:rooms_read	List the titles of rooms that you're in
spark:rooms_write	Manage rooms on your behalf
spark:memberships_read	List the people in rooms that you're in
spark:memberships_write	Invite people to rooms on your behalf
spark:messages_read	Read the content of rooms that you're in

spark:messages_write	Post and delete messages on your behalf
spark:teams_read	List the teams you are a member of
spark:teams_write	Manage teams on your behalf
spark:team_memberships_read	List the people in the teams that you are in
spark:team_memberships_write	Add people to teams on your behalf
spark:webhooks_read	See all webhooks created on your behalf
spark:webhooks_write	Modify or delete webhooks created on your behalf

The following sections examine some of the APIs you can use to create rooms, add people, and send messages in a room.

Organizations API

In Webex, an organization represents a group of users. Organizations can have hierarchical relationships where they manage other entities or are managed by them. Further technical details are available at https://developer.webex.com/docs/api/v1/organizations/.

Only an administrator can access the organization's API. Table 10-3 shows the methods for getting organizational details using the Organizations API.

Table 10-3 *Webex: Organization API*

Method	API	Description
GET	https://webexapis.com/v1/organizations	List all the organizations
GET	https://webexapis.com/v1/organizations/{orgId}	Get details about an organization

Teams API

A team is a group of people with rooms visible to all members. Please see the details: https://developer.webex.com/docs/api/v1/teams.

The Teams API manages teams—, creating, deleting, and renaming them. Table 10-4 list the various operations that can be performed on the Teams API.

Table 10-4 *Webex: Teams API*

Method	API	Description
GET	https://webexapis.com/v1/teams	List all teams
POST	https://webexapis.com/v1/teams	Create a new team
GET	https://webexapis.com/v1/teams/{teamId}	Get details about a particular team
PUT	https://webexapis.com/v1/teams/{teamId}	Update details about a team
DELETE	https://webexapis.com/v1/teams/{teamId}	Delete a team

For example, say that you want to use the Teams API to create a new DevNet Associate Certification Room team. To do so, you use the POST method and the API: https://webexapis.com/v1/teams.

You can use a Python request to make the REST call. Example 10-1 shows a Python script that sends a POST request to create a new team. It initializes variables such as the base URL, payload, and headers and calls the request.

Example 10-1 *Python Code to Create a New Team*

```
“”” Create Webex Team “””


import json

import requests


URL = "https://webexapis.com/v1/teams"

PAYLOAD = {

    "name": "DevNet Associate Certification Team"

}

HEADERS = {

    "Authorization": "Bearer

MDA0Y2VlMzktNDc2Ni00NzI5LWFiNmYtZmNmYzM3OTkyNjMxNmI0ND-

    VmNDktNGE1_PF84_consumer",

    "Content-Type": "application/json"

}

RESPONSE = requests.request(“POST”, URL, data=json.dumps(PAYLOAD),

headers=HEADERS)

print(RESPONSE.text)
```

Rooms API

Rooms are virtual meeting places where people post messages and collaborate to get work done. Please see the details: https://developer.webex.com/docs/api/v1/rooms.

The Rooms API manages rooms—to create, delete, and rename them. Table 10-5 lists the operations that can be performed with the Rooms API.

Table 10-5 *Webex: Rooms API*

Method	API	Description
GET	https://webexapis.com/v1/rooms	List all the rooms
POST	https://webexapis.com/v1/rooms	Create a new room
GET	https://webexapis.com/v1/rooms/{roomId}	Get room details
GET	https://webexapis.com/v1/rooms/{roomId}/meetingInfo	Get room meeting details
PUT	https://webexapis.com/v1/rooms/{roomId}	Update room details
DELETE	https://webexapis.com/v1/rooms/{roomId}	Delete a room

You can use the Rooms API to create a room. When you do, an authenticated user is automatically added as a room member. To create a room, you can use the POST method and the

API https://webexapis.com/v1/rooms.

Example 10-2 shows the Python request code that creates the DevAsc Team Room. It initializes variables such as the base URL, payload, and headers and calls the request. The header consists of the bearer token of the authenticated user or the integration, along with other parameters.

Example 10-2 *Python Request to Create a New Room*

```
“”” Create Webex Room “””
import json
import requests
import pprint

URL = "https://webexapis.com/v1/rooms"
PAYLOAD = {
    "title": "DevAsc Team Room"
}
HEADERS = {
    "Authorization": "Bearer
MDA0Y2VlMzktNDc2Ni00NzI5LWFiNmYtZmNmYzM3OTkyNjMxNmI0ND-
    VmNDktNGE1_PF84_consumer",
    "Content-Type": "application/json"
}
RESPONSE = requests.request(“POST”, URL, data=json.dumps(PAYLOAD),
headers=HEADERS)
pprint.pprint(json.loads(RESPONSE.text))
```

Example 10-3 shows the response to creating a room. The response includes the creation time, owner, and ID, which can be used in subsequent calls.

Example 10-3 *Response to the Successful Creation of a Room*

```
$ python3 CreateRoom.py
{'created': '2020-02-15T23:13:35.578Z',
 'creatorId': 'Y2lzY29zcGFyazovL3VzL1BFT1BMRS8wYWZmMmFhNC1mN2IyLTQ3MWU-
 tYTIzMi0xOTEyNDgwYmDEADB',
 'id':
'Y2lzY29zcGFyazovL3VzL1JPT00vY2FhMzJiYTAtNTA0OC0xMWVhLWJiZWItYmY1MWQyNGRm
 MTU0',
 'isLocked': False,
 'lastActivity': '2020-02-15T23:13:35.578Z',
 'ownerId': 'consumer',
 'title': 'DevAsc Team Room',
 'type': 'group'}
$
```

You can use the Rooms API to get a list of all the rooms created. To do so, use the GET method and the API at https://webexapis.com/v1/rooms.

Example 10-4 shows how to use the **curl** command to make the REST call. This script sends a GET request to list all rooms a particular user belongs to.

Example 10-4 ***curl*** *Script for Getting a List of All Rooms*

```
$ curl -X GET \
 https://webexapis.com/v1/rooms \
 -H 'Authorization: Bearer
DeadBeefMTAtN2UzZi00YjRiLWIzMGEtMThjMzliNWQwZGEyZTljNWQxZTktNTRl_PF84_1eb65fd
f-9643-417f-9974-ad72cae0e10f'
```

Memberships API

A membership represents a person's relationship to a room. Please see the details: https://developer.webex.com/docs/api/v1/memberships. You can use the Memberships API to list members of any room that you're in or create memberships to invite someone to a room. Memberships can also be updated to make someone a moderator or deleted to remove someone from the room. Table 10-6 list the operations that can be performed for the Memberships API, such as listing memberships and adding a new member.

Table 10-6 *Webex: Memberships API*

Method	API	Description
GET	https://webexapis.com/v1/memberships	List memberships
POST	https://webexapis.com/v1/memberships	Add a new member
GET	https://webexapis.com/v1/memberships/{membershipId}	Get details about a member
PUT	https://webexapis.com/v1/memberships/{membershipId}	Update details about a member
DELETE	https://webexapis.com/v1/memberships/{membershipId}	Delete a member

You can use the Memberships API to add a new member to a given room (that is, create a new membership) by using the POST method and the API https://webexapis.com/v1/memberships.

You can use Python to make a REST call. Example 10-5 show a **curl** script that sends a POST request to add a new member with the email ID newUser@devasc.com to the room.

Example 10-5 *Python Script to Add a New Member to a Room*

```
“”” Add new Member to a Webex Room “””


import json
import requests
import pprint


URL = "https://webexapis.com/v1/memberships"
PAYLOAD = {
    "roomId" : "Y2lzY29zcGFyazovL3VzL1JPT00vY2FhMzJiYTAtNTA0OC0xMWVhLWJiZ-
    WItYmY1MWQyNGRDEADB",
  "personEmail": "newUser@devasc.com",
  "personDisplayName": "Cisco DevNet",
  "isModerator": "false"
}
HEADERS = {
    "Authorization": "Bearer
MDA0Y2VlMzktNDc2Ni00NzI5LWFiNmYtZmNmYzM3OTkyNjMxNmI0ND-
    VmNDktNGE1_PF84_consumer",
    "Content-Type": "application/json"
}
RESPONSE = requests.request(“POST”, URL, data=json.dumps(PAYLOAD),
headers=HEADERS)
pprint.pprint(json.loads(RESPONSE.text))
```

Messages API

Messages are communications that occur in a room. In Webex, each message is displayed on its line, along with a timestamp and sender information. The Messages API can list, create, and delete messages.

Messages can contain plaintext, rich text, and a file attachment. Table 10-7 shows the API for sending messages to Webex.

Table 10-7 *Webex: Message API*

Method	API	Description
GET	https://webexapis.com/v1/messages	List messages
GET	https://webexapis.com/v1/messages/direct	List one-to-one message
POST	https://webexapis.com/v1/messages	Post a new message
GET	https://webexapis.com/v1/messages/{messageId}	Get details about a message
DELETE	https://webexapis.com/v1/messages/{messageId}	Delete a message

You can use the Messages API to add a new message to a given room (that is, create a new message). To do so, you use the POST method and the

API https://webexapis.com/v1/messages.

You can use a Python request to make a REST call. Example 10-6 shows a **curl** script that sends a POST message to add a new message to a particular room.

Example 10-6 *Python Script to Add a New Message to a Room*

```
“”” Send Webex Message “””


import json

import requests

import pprint


URL = "https://webexapis.com/v1/messages"

PAYLOAD = {

    "roomId" : "Y2lzY29zcGFyazovL3VzL1JPT00vY2FhMzJiYTAtNTA0OC0xMWVhLWJiZ-

    WItYmY1MWQyNGRmMTU0",

    "text" : "This is a test message"

}

HEADERS = {

    "Authorization": "Bearer

NDkzODZkZDUtZDExNC00ODM5LTk0YmYtZmY4NDI0ZTE5ZDA1MGI-

    5YTY3OWUtZGYy_PF84_consumer",

    "Content-Type": "application/json",

}

RESPONSE = requests.request(“POST”, URL, data=json.dumps(PAYLOAD),

headers=HEADERS)

pprint.pprint(json.loads(RESPONSE.text))
```

Bots, bot frameworks and tools

A *bot* (short for chatbot) is a code or an application that simulates a human conversation. Users communicate with a bot via the chat interface or voice, just as they would talk to a natural person. Bots help users automate tasks, bring external content into the discussion, and gain efficiency. Webex has a rich set of APIs that make it easy and simple for any developer to add a bot to any Teams room. In Webex, bots are similar to regular Webex users. They can participate in one-to-one and group spaces, and users can message or add them to a group space. A special badge is added to a bot’s avatar in the Webex clients so users know they’re interacting with a bot instead of a human.

A bot can only access messages sent to it directly. In group spaces, bots must be @mentioned to access a message. In one-to-one spaces, a bot has access to all messages from the user. Bots are typically of the three types described in Table 10-8.

Table 10-8 *Bot Types*

Type	Description
Notification bot	Events from external services are brought in and posted in Webex. Examples of events include build complete, retail inventory status, and the temperature today.
Controller bot	External systems with APIs allow third-party apps to be integrated to control them. For example, you could control turning lights on or off by invoking a lights bot.
Assistant bot	Virtual assistants usually understand natural language, so a user can ask questions of bots as they would ask humans (for example, "@Merakibot, how many wifi devices are currently on floor 2?")

Several bot frameworks can significantly simplify bot development by abstracting away the low-level communications with the Webex REST API, such as creating and sending API requests and configuring webhooks. You can focus on building the interaction and business logic of a bot. These are two popular bot frameworks:

- **Flint:** Flint is an open-source bot framework that supports regex pattern matching for messages.
- **Botkit:** Botkit is a popular open-source bot framework that offers advanced conversational support and integrations with a comprehensive array of natural language processing and storage providers.

One of the most significant starting points for learning about and creating your bots for Webex is the DevNet Code Exchange at https://developer.cisco.com/codeexchange/github/repo/howdyai/botkit, which is shown in Figure 10-8.

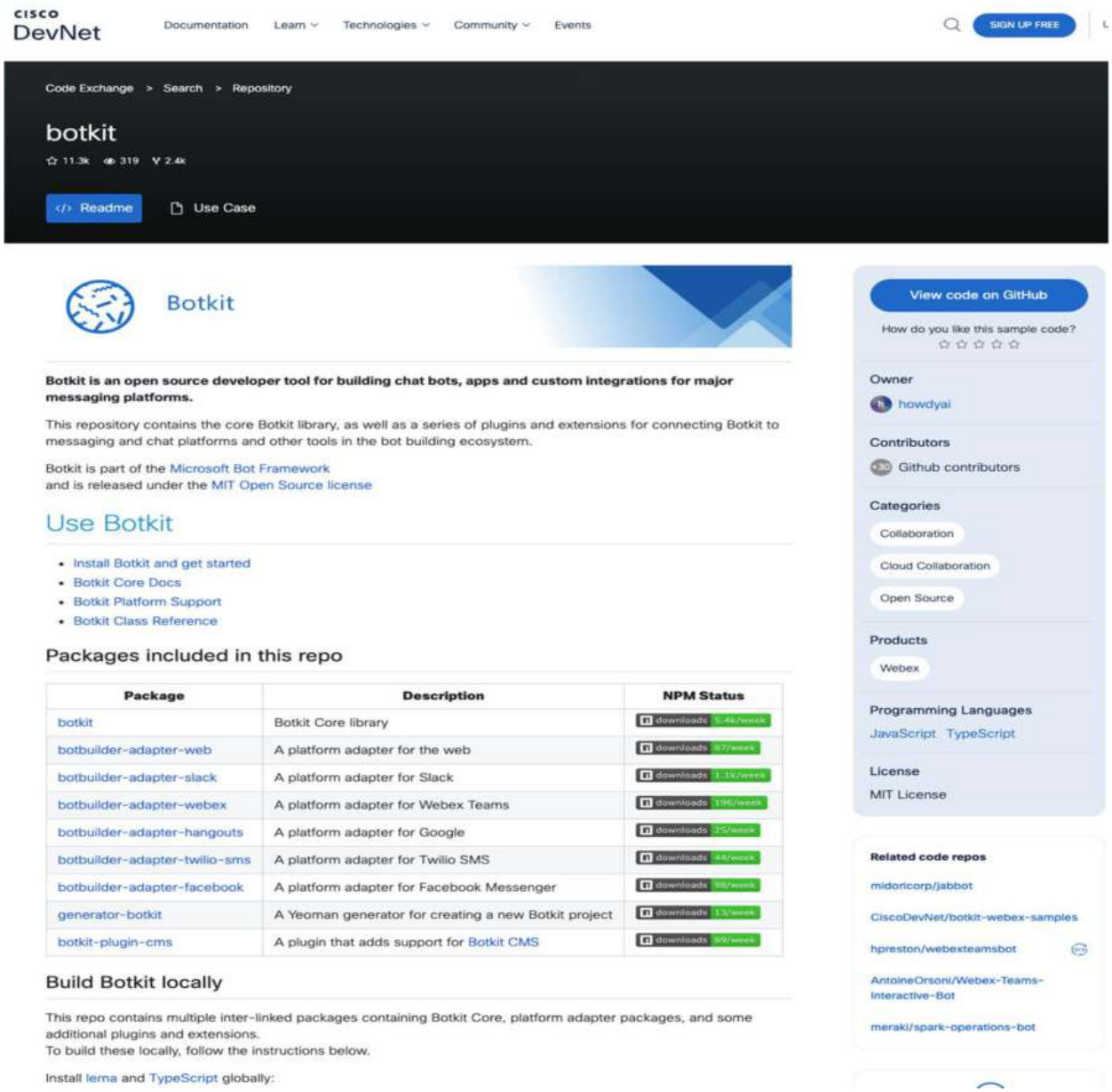


Figure 10-8 *DevNet Code Exchange: Building Your First Bot*

Guest Issuer

Guest issuer applications give guest users temporary access to users within the organization. Guest issuers can be created at https://developer.Webex.com/my-apps/new/guest-issuer. The only thing required to create a new guest issuer is the name. A new guest issuer ID and shared secret will be generated and can be used subsequently. The main reason for using a guest issuer is to interact with users who do not have a Webex account. These users might be visitors to a website who you'd like to message over Webex. Or they might be customers in a store with whom you'd like a video call. These guest users can interact with regular Webex users via tokens generated by a guest issuer application.

Guest users of Webex authenticate using guest tokens. Guest tokens use the JSON Web Token (JWT) standard to create and share authentication credentials between SDKs and widgets and the Webex REST API. These tokens are exchanged for an access authentication token that can be used for a limited time and for a limited purpose to interact with regular Webex users.

Each guest token should be associated with an individual user of an application. The guest's activity within Webex, such as message activity or call history, will persist, just as it would for a regular Webex user. While guest users can interact with regular Webex users, they cannot interact with other guests. Example 10-7 shows a Python code snippet that creates a JWT token from the guest issuer ID and secret and passes it in the authentication headers. Using any of the APIs to interact with other users in the system is possible.

Example 10-7 *Python Code to Generate a JWT Token for a Guest Issuer*

```
""" Generate JWT """


import base64
import time
import math
import jwt


EXPIRATION = math.floor(time.time()) + 3600 # 1 hour from now
PAYLOAD = {
    "sub": "devASC",
    "name": "devASC-guest",
    "iss": "GUEST_ISSUER_ID",
    "exp": EXPIRATION
}


SECRET = base64.b64decode('GUEST_ISSUE_SECRET')


TOKEN = jwt.encode(PAYLOAD, SECRET)


print(TOKEN.decode('utf-8'))
HEADERS = {
    'Authorization': 'Bearer ' + TOKEN.decode('utf-8')
}
```

Webex SDKs

As of this writing, there is a variety of SDKs available; some of them are official Webex SDKs, and others are from the community. The following is a selection of the Web Teams SDKs that are available:

- **Go (go-cisco-Webex-teams):** A Go client library (by jbogarin)
- **Java (spark-java-sdk):** A Java library for consuming the RESTful APIs (by Cisco Webex)
- **Node.js (https://github.com/webex/webex-js-sdk):** A collection of Node.js modules targeting the REST API (by Cisco Webex)
- **PHP (SparkBundle):** A Symfony bundle (by CiscoVE)
- Python (https://github.com/WebexCommunity/WebexPythonSDK): An SDK that works with the REST APIs in native Python

The following are some advanced APIs:

- **SDK for Android:** Integrates messaging and calling into Android apps (by Cisco Webex)
- **SDK for Browsers:** Integrates calling into client-side JavaScript applications (by Cisco Webex)
- **SDK for iOS:** Integrates messaging and calling into iOS apps (by Cisco Webex)
- **SDK for Windows:** Integrates messaging and calling into Windows apps (by Cisco Webex)
- **Widgets:** Provides components that mimic the web user experience (by Cisco Webex)

Cisco Finesse

The Cisco Finesse desktop is a call agent and supervisor desktop solution designed to meet the growing needs of agents, supervisors, and the administrators and developers who support them. The Cisco Finesse desktop runs in a browser, which means you install Cisco Unified Contact Center Express (Unified CCX), and agents start by simply typing in the URL for the Unified CCX server. The desktop is more than an agent state and call-control application. It is an OpenSocial gadget container with third-party applications in a single-agent desktop experience. Rather than switching between applications, agents have easy access to

all applications and tools from a single window, which increases their efficiency. Figure 10-9 shows the architecture and high-level flow of Finesse, which involves the following steps:

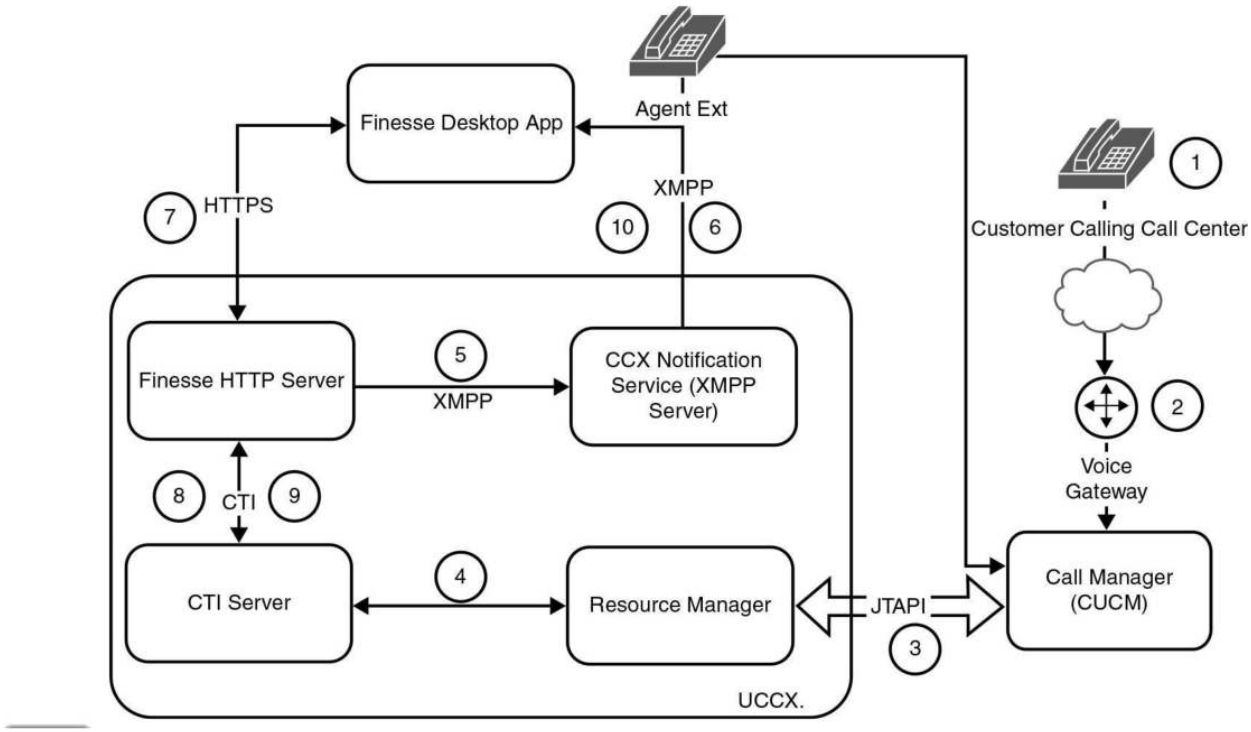


Figure 10-9 *Finesse High-Level Flow*

Key Topic

Step 1. The call arrives from either the PSTN or a VoIP connection to the gateway.

Step 2. The gateway then hands the call to Unified CM, which invokes the preregistered application. In this case, it is handled by Unified CCX.

Step 3. Unified CM notifies the Unified CCX about the incoming call via JTAPI.

Step 4. After consulting the resource manager (routing the call to the agent based on skills, priority, and rebalancing), Unified CCX notifies Finesse via a computer telephony integration (CTI) connection.

Step 5. Finesse performs internal processing and then publishes a notification to Notification Service.

Step 6. The Finesse desktop receives this notification from Notification Service via the Bidirectional-streams Over Synchronous HTTP (BOSH) connection (Extensible Messaging and Presence Protocol [XMPP]).

Step 7. The agent requests to perform an operation (such as answering a call); the app makes this HTTP (REST) request to Finesse Web Services.

Step 8. Finesse processes the request and then, if necessary, requests action with Unified CCE via CTI connection. Where applicable, Unified CCE performs/forwards the requested action.

Step 9. Unified CCE notifies Finesse via CTI connection about whether the request was successful or caused an error.

Step 10. Finesse processes the notification (successful/error) and publishes it to the Notification Service. The agent web browser receives this notification from the Notification Service via the BOSH connection.

The Finesse agent goes through various states about the agent workflow (see Table 10-9).

Table 10-9 *User States in Finesse*

State	Description
LOGIN	The agent is signing in to the system. This is an intermediate state.
LOGOUT	The agent is signed off the system.
READY	The agent is ready to take calls.
NOT_READY	The agent is signed in but has yet to be ready to take calls. The agent could be on a break, the shift might be over, or the agent might be in between calls.
RESERVED	This is transient, as the agent gets chosen but has yet to answer the call.
TALKING	The agent is on a call.
HOLD	The agent puts the call on hold.

Cisco Finesse API

The Cisco Finesse API is a modern, open-standards-based web API, exposed via REST. Each function in the Cisco Finesse user interface has a corresponding REST API that allows developers to use all integrations. The API's extensibility and ease of use are unprecedented on Unified CCX. Agents and supervisors use the Cisco Finesse desktop APIs to communicate between the Finesse desktop and Finesse server, and they use Unified Contact Center Enterprise (Unified CCE) or Unified Contact Center Express (Unified CCX) to send and receive information.

The Finesse APIs can be broadly classified into the following categories:

- User
- Dialog
- Queue
- Team
- ClientLog
- Task Routing APIs
- Single Sign-On
- TeamMessage

Cisco Finesse supports both HTTP and HTTP Secure (HTTPS) client requests. Cisco Finesse desktop operations can be performed using one of the many available REST-like HTTP/HTTPS requests. Operations on specific objects are performed using the object's ID in the REST URL. For example, the URL to view a single object (HTTP) would be as follows:

http://<FQDN>:<port>/finesse/api/<object>/<objectID>

where FQDN is the fully qualified domain name of the Finesse server.

Finesse configuration APIs require the application user ID and password to be established during installation for authentication purposes.

Finesse APIs use the following HTTP methods to make requests:

- **GET:** Retrieves a single object or list of objects (for example, a single user or list of users).
- **PUT:** Replaces a value in an object (for example, to change the state of a user from NOT_READY to READY).
- **POST:** Creates a new entry in a collection (for example, to create a new reason code or wrap-up reason).
- **DELETE:** This function removes an entry from a collection (for example, to delete a reason code or wrap-up reason).

Finesse uses the standard HTTP status codes (for example, 200, 400, and 500) in the response to indicate a request's overall success or failure.

API Authentication

All Finesse APIs use HTTP BASIC authentication, which requires the credentials to be sent in the authorization header. The credentials contain the username and password, separated by a single colon (:), within a Base64-encoded string. For example, the authorization header would contain the following string:

```
"Basic ZGV2YXNjOnN0cm9uZ3Bhc3N3b3Jk"
```

where ZGV2YXNjOnN0cm9uZ3Bhc3N3b3Jk is the Base64-encoded string *devasc*:*strongpassword* (where *devasc* is the username, and *strongpassword* is the password). Example 10-8 shows three lines of code that do Base64 encoding to plug the value in the authorization headers.

Example 10-8 *Python Code to Generate Base64 Encoding*

```
""" Generate Base64 Encoding """
import base64
ENCODED = base64.b64encode('devasc:strongpassword'.encode('UTF-8'))
print(ENCODED.decode('utf-8'))
```

With Single Sign-On mode, the authorization header would contain the following string:

```
"Bearer <authtoken>"
```

Finesse User APIs

Table 10-10 lists the various methods and User APIs to perform operations with the user, such as listing, logging in, and changing properties.

Table 10-10 *Finesse User APIs*

Method	API	Description
GET	http://<FQDN>/finesse/api/User/<id>	Get a copy of the user object
GET	http://<FQDN>/finesse/api/User	Get a list of all users
PUT	http://<FQDN>/finesse/api/User/<id> with XML body: <User> <state>LOGIN</state> <extension>5250001</extension> </User>	Sign in to the CTI server
PUT	http://<FQDN>/finesse/api/User/<id> with XML body: <User> <state>READY</state> </User>	Set the user's state: ■ READY ■ NOT_READY ■ LOGOUT
GET	http://<FQDN>/finesse/api/User/<id>/PhoneBooks	Get a list of phone books and the first 1500 associated contacts for that user

A full list of all User state change APIs with details can be found at https://developer.cisco.com/docs/finesse/#!userchange-agent-state/userchange-agent-state.

For example, the User—Sign in to Finesse API forces you to sign in. Say that you use the following information with this API:

- **Finesse server FQDN:** http://hq-uccx01.abc.inc
- **Agent name:** Anthony Phyllis
- **Agent ID:** user001
- **Agent password:** cisco1234

Example 10-9 shows a simple call using Python requests. The API call for user login uses the PUT request and an XML body that sets the state to LOGIN.

Example 10-9 *Python Request: Finesse User Login*

```
“”” Finesse - User Login”””
import requests


URL = "http://hq-uccx.abc.inc:8082/finesse/api/User/Agent001"
PAYLOAD = (
    "<User>" +
    "    <state>LOGIN</state>" +
    "    <extension>6001</extension>" +
    "</User>"
)


HEADERS = {
    'authorization': "Basic QWdlbnQwMDE6Y2lzY29wc2R0",
    'content-type': "application/xml",
    }
RESPONSE = requests.request("PUT", URL, data=PAYLOAD, headers=HEADERS)
print(RESPONSE.text)
print(RESPONSE.status_code)
```

As another example, the User State Change API lets the users change their state. State changes could be any one as shown in Table 10-9. Say that you use the following information with this API:

- **Finesse server FQDN:** http://hq-uccx01.abc.inc
- **Agent name:** Anthony Phyllis
- **Agent ID:** user001
- **Agent password:** cisco1234

The API changes the state to READY.

Example 10-10 shows a simple call using Python requests. The API call for user login uses the PUT request along with an XML body that sets the state to READY.

Example 10-10 *Python Request for a Finesse User State Change*

```
“”” Finesse - User State Change”””
import requests


URL = "http://hq-uccx.abc.inc:8082/finesse/api/User/Agent001"
PAYLOAD = (
    "<User>" +
    "    <state>READY</state>" +
    "</User>"
)


HEADERS = {
    'authorization': "Basic QWdlbnQwMDE6Y2lzY29wc2R0",
    'content-type': "application/xml",
    }
RESPONSE = requests.request(“PUT”, URL, data=PAYLOAD, headers=HEADERS)
print(RESPONSE.text)
print(RESPONSE.status_code)
```

Finesse Team APIs

The Team object represents a team and contains the URI, the team name, and the users associated with the team. Table 10-11 shows the Finesse Team APIs to access the Team object and list all team messages.

Table 10-11 *Finesse Team APIs*

Method	API	Description
GET	http://<FQDN>/finesse/api/Team/<id>?includeLoggedOutAgents=true	Allow a user to get a copy of the Team object
GET	http://<FQDN>/finesse/api/Team/<teamid>/TeamMessages	Get a list of all active team messages for a particular team

A full list of the Team APIs, with details, can be found at https://developer.cisco.com/docs/finesse/#team-apis.

The Python script in Example 10-11 shows how to make an API call to get details about Team ID 2.

Example 10-11 *Python Request to Get Finesse Team Details*

```
import requests
url = "https://hq-uccx.abc.inc:8445/finesse/api/Team/2"
headers = {
     'authorization': "Basic QWdlbnQwMDE6Y2lzY29wc2R0",
      'cache-control': "no-cache",
      }
response = requests.request("GET", url, headers=headers)
print(response.text)
```

Dialog APIs

The Dialog object represents a dialog (voice or non-voice) between two or more users. There are many flavors of dialog APIs. Table 10-12 shows the Finesse dialog API, which lets two users make a call with each other.

Table 10-12 *Sample of Finesse Dialog APIs*

Method	API	Description
POST	http://<FQDN>/finesse/api/User/<id>/Dialogs with XML Body: <Dialog> <requestedAction>MAKE_CALL</requestedAction> <fromAddress>6001</fromAddress> <toAddress>6002</toAddress> </Dialog>	Allow a user to make a call
PUT	http://<FQDN>/finesse/api/Dialog/<dialogId> with XML body: <Dialog> <requestedAction>START_RECORDING</requestedAction> <targetMediaAddress>6001</targetMediaAddress> </Dialog>	Allow a user to start recording an active call

A full list of the Dialog APIs, with details, can be found at https://developer.cisco.com/docs/finesse/#dialog-apis.

The Python script in Example 10-12 shows how to make an API call to make a call between extension 6001 and extension 6002.

Example 10-12 *Python Request to Initiate a Dialog Between Two Numbers*

```
“”” Finesse - Initiate a dialog between two numbers “””
import requests


URL = "http://hq-uccx.abc.inc:8082/finesse/api/User/Agent001/Dialogs"
PAYLOAD = (
    "<Dialog>" +
    "    <requestedAction>MAKE_CALL</requestedAction>" +
    "    <fromAddress>6001</fromAddress>" +
    "    <toAddress>6002</toAddress>" +
    "</Dialog>"
)


HEADERS = {
    'authorization': "Basic QWdlbnQwMDE6Y2lzY29wc2R0",
    'content-type': "application/xml",
    'cache-control': "no-cache",
    }
RESPONSE = requests.request(“POST”, URL, data=PAYLOAD, headers=HEADERS)
print(RESPONSE.text)
print(RESPONSE.status_code)
```

Finesse Gadgets

As this chapter indicates, the Finesse desktop application is an OpenSocial gadget container. This means that an agent or anyone else can customize what is on the desktop. Gadgets are built using HTML, CSS, and JavaScript. A gadget is a simple but powerful tool that allows an agent to quickly send one or more web pages (carefully curated by the client teams themselves) to their caller over Instant Messenger or via email. This feature automates the four or five manual steps usually associated with this task.

A gadget is defined using declarative XML syntax that is processed by a gadget server so that it can be embedded into the following contexts:

- Standalone web pages
- Web applications
- Other gadgets

The Finesse JavaScript library can be found on a Finesse server at http(s)://<FQDN>:<port>/desktop/assets/js/doc/index.html.

Several gadgets are available at https://developer.cisco.com/docs/finesse/#!sample-gadgets/sample-gadgets.

Webex Meetings APIs

The Webex Meetings APIs let users incorporate Cisco Webex meetings into their applications:

- **URL API:** The URL API is a convenient, lightweight, HTTP/HTTPS-based mechanism that provides browser-based external hooks into Webex Meetings services. It is typically used in enterprise portal integrations to support basic interactions such as single sign-on (SSO), scheduling meetings, starting and joining simple meetings, and inviting attendees and presenters. Figure 10-10 provides a high-level overview of where the URL API integration layer resides.
- **XML API:** If more advanced integration is needed than is possible with the URL API, Cisco strongly recommends using the Webex Meetings XML API. The XML API is a comprehensive set of services that supports most aspects of Webex Meetings services, including detailed user management, comprehensive scheduling features, and attendee management and reporting. The Webex XML API uses a service-oriented architecture (SOA) to provide comprehensive services to external applications wishing to interact with one or more Webex services.
- **Teleconference Service Provider (TSP) API:** The TSP API provides full-featured XML-based integration with audio conferencing networks. The integration architecture supports redundancy, failover, and load balancing, and it provides robust features that tightly integrate audio conferencing capabilities with telephony management and provisioning in the Webex environment.

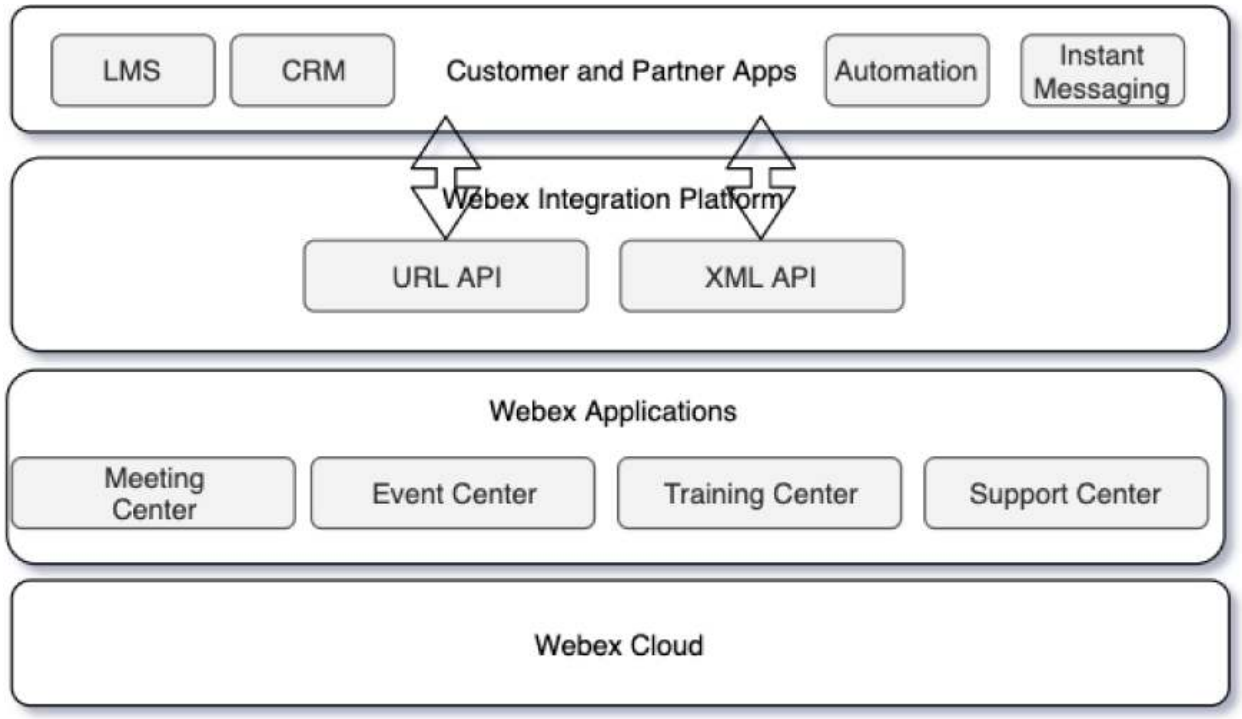


Figure 10-10 *Webex Meetings API Architecture*

Table 10-13 summarizes the available URL API services and simple commands for integrating with a Webex Meetings–hosted website.

Table 10-13 *Webex Meetings Services Supported via APIs*

Service Name	Integration Usage
Managing User Accounts	Creating a new user account Editing an existing user account Activating and deactivating user accounts
Webex-hosted website login/logout	Using an authenticated server-to-server connection for logging in to and out of a Webex Meetings–hosted website
Using and Managing Meetings	Scheduling a meeting Editing a meeting Starting or deleting a host's scheduled meeting Listing all scheduled meetings Listing all open meetings Joining an open meeting
Modifying My Webex	Modifying user information on the My

Meetings page	Webex Meetings page Managing a user's My Contacts list
Using Attendee Registration forums	Creating a registration form Determining the current required, optional, or do-not-display settings for a registration page Adding check boxes, buttons, and drop-down lists to a registration form
Managing Attendee Lists	Adding attendees to a list of invited users Removing attendees from a list of invited users
Playing back a recorded event	Allowing an attendee to get a list of recorded events for playback
Querying for Questions and Answers	Viewing a list of custom questions created by the host Viewing attendees' answers to custom questions Viewing a list of standard questions created by the host Viewing attendees' answers to standard questions

Making Recording Training Sessions available for viewing	Making all aspects of a previously recorded Training Center session available for later playback
Reporting	Sending email notifications with attendee information Displaying reporting information about training sessions you hosted Viewing a list of enrollees and attendees for events Viewing a list of all events a specific attendee has joined Viewing a list of people who have downloaded files

Authentication

There are three methods by which a user can interact with the Webex Meetings via APIs:

- **Webex Meetings integration:** An application can authenticate XML API requests using OAuth 2.0 access tokens. This authentication method does require end-user authorization and authentication and is best used when performing API functions on behalf of the active user.
- **Administrative account:** A system account can perform administrative functions on behalf of host users. This method requires the username and password to be passed via XML.
- **User accounts:** A system account can perform functions only for its own accounts. This method requires the username and password to be passed via XML.

Integration API Keys

In order to use the Webex Meetings APIs, you need to preregister an API key and then use the HM256 algorithm to generate a JWT token to access a Webex anonymous API (e.g., **GetSessionInfoAgg**, **GetAllSitesByEmailAgg**).

To apply for an API key, visit https://api.Webex.com/gapi/registerapikey. After you apply for the key, you get the API and the secret (see Example 10-13).

Example 10-13 *Webex Meeting Integration Key Obtained via the Portal*

```
Name: myname mylastname
Email: myname@devasc.com
New API key: deadbeaf-d1e3-4000-993f-0d9479ab4944
New Passcode: 2BEEF
New Secret key:
DEADBEEFNYH54RP8THOJSXM8OJKOUJVHXKGK7QQZSL1KF2D967JY5XZIR64B3GY5NGRWMBVXG132X
DGZOT5XJ62VQU4558I13J8IQ7TPEIPFB9MNG48WJM67F5C82P01VYTQTT4428T7RYQN0YMSHXAX8A
RLVQHHPNCCH1RJZYRIUF3AI0O0A6BA3BQN54DJ9VN3V9XPKUZZM1E570X07IKTSHDVTPW8CWMU5Y9
XZG76Z9FFA5UN1DJ7N39RXQQE99TRFA5HDBL2BJBPUOX6NMY6BDBZ4JP2IQ3RP3D6D7CUBP5U7W5E
KBWVTRZQXBY0L4M98Y1BR9HX5FD9D9HWYA
Expiration time: 4/27/20 6:22 AM GMT
```

Webex XML APIs

The Webex XML model utilizes the exchange of well-formed documents to deploy messages. Each message relates to a specific Webex operation. An XML request document specifies the desired action of the particular service's elements. The XML response document returned by the Webex XML server describes the revised state of that service, as determined by the Webex XML server.

Figure 10-11 provides a simplified representation of the architecture of the Webex model.

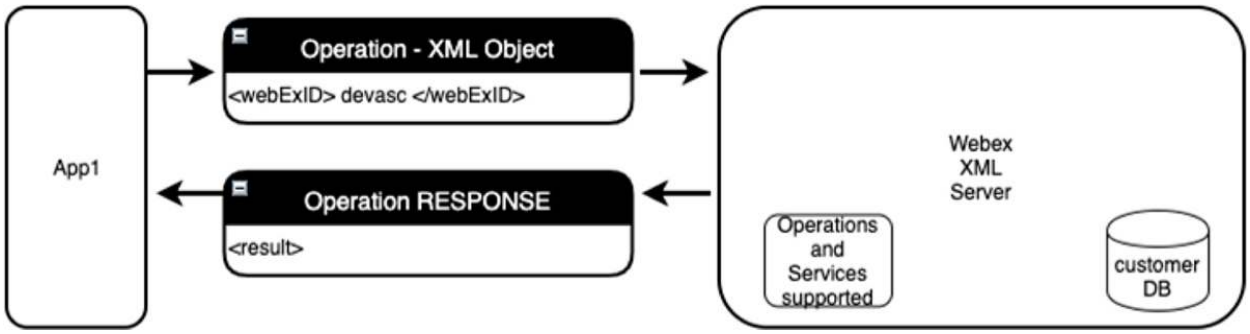


Figure 10-11 *Webex XML Model Architecture*

Key Topic

Figure 10-11 shows that an application sends a request XML document to perform a specific request—, such as creating a new meeting—to the Webex XML server. This request XML document describes the state of the elements associated with the operation for this request. The Webex XML server then returns the revised element values for this operation to the original application via a response XML document created by the Webex XML document processor.

The Webex XML server can be accessed at https://api.Webex.com/WBXService/XMLService.

Creating a New Meeting

The CreateMeeting API enables users to schedule a meeting and returns a unique meeting key for the session. Table 10-14 show the API and the XML body for creating a new meeting.

Table 10-14 *Creating or Scheduling a Webex Meeting*

Method	API	Description
POST	https://api.Webex.com/WBXService/XMLService with XML body: <body> <bodyContent xsi:type="java:com.Webex.service.binding.meeting.CreateMeeting"> <metaData> <confName>Branding Meeting</confName> </metaData> <schedule> <startDate/> </schedule> </bodyContent> </body>	Create a new meeting with various attributes such as name, start time, password, and attendees

Example 10-14 shows how to make an API call to create a new meeting with a particular subject line. The response is usually the new meeting ID.

Example 10-14 *Creating a New Meeting Using the CreateMeeting API*

```
curl -X POST \
  https://api.Webex.com/WBXService/XMLService \
  -H 'cache-control: no-cache' \
  -H 'content-type: application/xml' \
  -d '<?xml version="1.0" encoding="UTF-8"?>
<serv:message xmlns:xsi="http://www.w3.org/2001/XMLSchema-instance">
    <header>
        <securityContext>
            <WebexID>devasc</WebexID>
            <password>Kc5Ac4Ml</password>
            <siteName>apidemoeu</siteName>
        </securityContext>
    </header>
    <body>
        <bodyContent xsi:type="java:com.Webex.service.binding.meeting.
        CreateMeeting">
            <metaData>
                <confName>Branding Meeting</confName>
            </metaData>
            <schedule>
                <startDate/>
            </schedule>
        </bodyContent>
    </body>
</serv:message>'
```

Listing All My Meetings Meeting

The LstsummaryMeeting API lists summary information for scheduled meetings. This API returns a unique meeting key for a session. Table 10-15 shows the XML data that needs to be sent in order to list all meetings sorted by start time.

Table 10-15 *Listing All Webex Meetings That Match Certain Criteria*

Method	API	Description
POST	https://api.Webex.com/WBXService/XMLService with XML body: <body> <bodyContent xsi:type="java:com.Webex.service.binding.meeting.LstsummaryMeeting"> <order> <orderBy>STARTTIME</orderBy> </order> </bodyContent> </body>	List all meetings that meet certain criteria

Example 10-15 shows how to make an API call to list meetings sorted by time.

Example 10-15 *Listing All Meetings for a User by Using the LstsummaryMeeting API*

```
curl -X POST \
  https://api.Webex.com/WBXService/XMLService \
  -H 'cache-control: no-cache' \
  -H 'content-type: application/xml' \
  -d '<serv:message xmlns:xsi="http://www.w3.org/2001/XMLSchema-instance">
  <header>
    <securityContext>
      <WebexID>devasc</WebexID>
      <password>Kc5Ac4Ml</password>
      <siteName>apidemoeu</siteName>
    </securityContext>
  </header>
  <body>
    <bodyContent
      xsi:type="java:com.Webex.service.binding.meeting.LstsummaryMeeting">
      <order>
        <orderBy>STARTTIME</orderBy>
      </order>
    </bodyContent>
  </body>
</serv:message>’
```

Setting or Modifying Meeting Attributes

The SetMeeting API enables hosts/users to update the information for a scheduled meeting that they are able to edit. Table 10-16 shows the XML data that needs to be sent in order to modify meeting attributes.

Table 10-16 *Modify Meeting Attributes*

Method	API	Description
POST	https://api.Webex.com/WBXService/XMLService with XML body: <bodyContent xsi:type="java:com.Webex.service.binding.meeting.SetMeeting"> <meetingkey>625579604</meetingkey> <participants> <attendees> <attendee> <person> <email>student@devasc.com</email> </person> </attendee> </attendees> </bodyContent>	Update meeting attributes such as attendees, record meeting, schedules, and media support

Example 10-16 shows how to make an API call to add a new attendee (student@devasc.com) to the meeting and send an email invitation to that person.

Example 10-16 *Adding a New User to a Meeting by Using the SetMeeting API*

```
curl -X POST \
  https://api.Webex.com/WBXService/XMLService \
  -H 'cache-control: no-cache' \
  -H 'content-type: application/xml' \
  -d '<serv:message xmlns:xsi="http://www.w3.org/2001/XMLSchema-instance">
  <header>
    <securityContext>
     <WebexID>devasc</WebexID>
      <password>Kc5Ac4Ml</password>
      <siteName>apidemoeu</siteName>
    </securityContext>
  </header>
 <body>
    <bodyContent
      xsi:type="java:com.Webex.service.binding.meeting.SetMeeting">
      <meetingkey>625579604</meetingkey>
      <participants>
        <attendees>
          <attendee>
            <person>
              <email>student@devasc.com</email>
            </person>
          </attendee>
        </attendees>
      </participants>
      <attendeeOptions>
        <emailInvitations>true</emailInvitations>
      </attendeeOptions>
      <schedule>
        <openTime>300</openTime>
      </schedule>
    </bodyContent>
  </body>
</serv:message>’
```

Deleting a Meeting

The DelMeeting API allows hosts to delete a meeting that is not currently in progress. The API continues to use the POST method, but the XML data contains the operation of deleting the meeting. Table 10-17 shows the XML data that needs to be sent in order to delete a meeting.

Table 10-17 *Deleting a Webex Meeting Using the Meeting Key*

Method	API	Description
POST	https://api.Webex.com/WBXService/XMLService with XML body: <body> <bodyContent xsi:type="java:com.Webex.service.binding.meeting.DelMeeting"> <meetingKey>625579604</meetingKey> </bodyContent> </body>	Delete or cancel a meeting, given the meeting ID

Example 10-17 shows how to make an API call to delete the meeting with ID =625579604.

Example 10-17 *Deleting a Meeting Using the DelMeeting API*

```
curl -X POST \
  https://api.Webex.com/WBXService/XMLService \
  -H 'cache-control: no-cache' \
  -H 'content-type: application/xml' \
  -d '<serv:message xmlns:xsi="http://www.w3.org/2001/XMLSchema-instance">
    <header>
        <securityContext>
            <WebexID>devasc</WebexID>
            <password>Kc5Ac4Ml</password>
            <siteName>apidemoeu</siteName>
        </securityContext>
    </header>
    <body>
        <bodyContent
            xsi:type="java:com.Webex.service.binding.meeting.DelMeeting">
            <meetingKey>625579604</meetingKey>
        </bodyContent>
    </body>
</serv:message>'
```

The Meetings XML API provides a method to integrate Webex Meetings services with your custom web portal or application. For more details about the Webex Meetings APIs, see https://developer.cisco.com/docs/Webex-meetings/#!Webex-meetings.

Webex REST API

The Webex Meetings RESTful API Suite offers developers a powerful tool for seamlessly integrating Webex functionality into their applications. By leveraging these APIs, you can create custom solutions that enhance collaboration, streamline workflows, and provide a more personalized meeting experience.

Core API Features:

Meeting Management: Schedule, list, update, and delete meetings programmatically.

Invitee Management: Add, remove, and manage meeting participants.

Recording Management: Retrieve and delete recordings for easy access and storage.

User Preference Customization: Allow users to tailor their Webex experience to their specific needs.

The Webex Meetings API Suite offers a comprehensive set of methods for managing meetings and their lifecycles. Here's a breakdown of some essential methods:

Creating a Meeting:

Method: **POST**

URL: https://webexapis.com/v1/meetings

Description:

Schedule a new meeting by providing details like topic, date, time, agenda, and participants.

Retrieving Meeting Information:

Method: **GET**

URL:

https://webexapis.com/v1/meetings/{meetingId} (Get details of a specific meeting)

https://webexapis.com/v1/meetings (List all meetings)

Description:

Retrieve details of a specific meeting by providing its unique identifier (meetingId).

List all meetings associated with your account using optional filters (e.g., date range).

Managing Meetings:

Method:

PATCH - https://webexapis.com/v1/meetings/{meetingId} (Partial Update)

PUT - https://webexapis.com/v1/meetings/{meetingId} (Full Update)

DELETE - https://webexapis.com/v1/meetings/{meetingId} (Delete)

Description:

Update specific aspects of a meeting (e.g., time, participants) using the PATCH method.

Replace all meeting details with a new set of information using the PUT method.

Permanently delete a meeting using the DELETE method.

Joining a Meeting:

Method: **POST**

URL: https://webexapis.com/v1/meetings/join

Description: Initiate a user's direct participation in a meeting using its unique identifier or other relevant information.

Additional Notes: The {meetingId} placeholder in the URLs represents the unique identifier for a specific meeting.

For a complete list of available methods, parameters, and response formats, refer to the official Webex Meetings API documentation - https://developer.webex.com/docs/api/v1/meetings

These methods provide a solid foundation for managing your Webex meetings programmatically and integrating Webex functionality into your applications.

Webex Devices

Webex Devices enable users to communicate and work with each other in real-time. The collaboration devices can be placed in meeting rooms or at desks. These devices are fully programmable except for the high-quality audio and video support. Through embedded APIs (often referenced as xAPI), you can extend and leverage Webex Room and Desk device capabilities in several ways:

- Create user interface components to customize the meeting experience and control IoT devices such as lights, curtains, and projectors from the device controls interface
- Configure devices at scale via automation scripts
- Initiate or respond to calls automatically
- Deploy code onto the devices without needing to deploy external control systems

Webex Devices includes the following devices, as shown in Figure 10-12:

- **Webex Board:** Cisco Webex Room Kit, Room Kit Plus, and Room Kit Pro
- **Room Devices:** Room 55, Room 70, Room 70 Dual, Board 55/55S, Board 70/70S, Board 85S, Cisco TelePresence MX200 G2, MX300 G2, MX700, MX800, MX800 Dual, SX10, SX20, and SX80
- **Webex Desk Device:** Cisco Webex DX80 and DX70 Collaboration Endpoint Software version 9

Figure 10-12 *Webex Boards, Room, and Desk Devices*

xAPI

xAPI is the API for collaboration endpoint software (Cisco CE and RoomOS) for both on-premises registered video conferencing devices (Unified CM and VCS) and devices registered to Cisco's cloud service (Cisco Webex Devices). xAPI consists of four major groups:

- Commands
- Configurations
- Status
- Events

Table 10-18 shows the high-level API categories that xAPI supports.

Table 10-18 *xAPI Categories*

Method	API	Description
GET	http://<ip-address>/status.xml	Get the complete status of the device
GET	http://<ip-address>/configuration.xml	Get the complete configuration of the device
GET	http://<ip-address>/command.xml	Get the complete command set supported by the device
GET	http://<ip-address>/valuespace.xml	Get an overview of all the value spaces used in the system settings, status information, and commands
POST	http://<ip-address>/putxml	Configure any settings on the device

xAPI Authentication

Access to xAPI requires the user to authenticate using HTTP basic access authentication as a user with the ADMIN role. Unauthenticated requests prompt a 401 HTTP response containing a basic access authentication challenge.

Suppose an application will issue multiple commands through xAPI. In that case, it is recommended that you use session authentication because standard basic authentication requires full re-authentication per request, which may affect your application's performance.

xAPI Session Authentication

Authenticating with your username and password combination for each API request might introduce too much latency for some use cases. To mitigate this, the API supports a session-based authentication mechanism.

To open a session, issue a POST to http://<ip-address>/xmlapi/session/begin with basic access authentication. The response sets a SessionId-cookie that can be used with subsequent requests.

Note that when using API session authentication, it is crucial to explicitly close the session when you are done. Please do so to avoid the device running out of sessions, as a limited number of concurrent sessions are available, and they do not time out automatically.

Creating a Session

You create a session by sending an xAPI session request to the endpoint. Example 10-18 shows a simple Python POST request, to which the server responds with the session cookie.

Example 10-18 *Python Script to Get a Session Cookie*

```
“”” Webex Devices - Get Session Cookie “””

import requests

URL = “http://10.10.20.159/xmlapi/session/begin”

HEADERS = {

     ‘Authorization’: “Basic ZGV2YXNjOkMxc2NvITIz”

      }


RESPONSE = requests.request(“POST”, URL, headers=HEADERS)

print(RESPONSE.headers[“Set-Cookie”])
```

The corresponding response header contains the session cookie, as shown in Example 10-19.

Example 10-19 *Response Header Containing the Session Cookie*

```
$ python3 sess.py

SessionId=031033bebe67130d4d94747b6b9d4e4f6bd2965162ed542f22d32d75a9f238f9;

Path=/; HttpOnly
```

Getting the Current Device Status

After a session is established, the session cookie is used in each interaction. Example 10-20 show how to use the cookie to get the device's current status by issuing a simple Python GET request.

Example 10-20 *Python Script to Get Endpoint Status*

```
“”” Webex Device - Endpoint Status “””

import requests
URL = “http://10.10.20.159/status.xml”
HEADERS = {
      ‘Cookie’:
“SessionId=c6ca2fc23d3f211e0517d4c603fbe4205c77d13dd6913c7bc12eef4085b7637b”
}

RESPONSE = requests.request(“GET”, URL, headers=HEADERS)
print(RESPONSE.text)
```

Setting Device Attributes

When a session is established, the session cookie is used in each interaction. Example 10-21 show how to use the cookie to set the camera position with certain pan and tilt values.

Example 10-21 *Python Script to Send a Command to Set the Camera Position*

```
“”” Webex Device - Set Camera Position “””

import requests

URL = "http://10.10.20.159/put.xml"

PAYLOAD = (
    '<Command>' +
    '  <Camera>' +
    '    <PositionSet command="True">' +
    '    <CameraId>1</CameraId>' +
    '    <Pan>150</Pan>' +
    '    <Tilt>150</Tilt>' +
    '    </PositionSet>' +
    '  </Camera>' +
    '</Command>'
)

HEADERS = {
    'Content-Type': "application/xml",
    'Cookie':
"SessionId=c6ca2fc23d3f211e0517d4c603fbe4205c77d13dd6913c7bc12eef4085b
7637b"
}

RESPONSE = requests.request(“POST”, URL, data=PAYLOAD, headers=HEADERS)
print(RESPONSE.text)
```

Registering an Event Notification Webhook

You can get a device or an endpoint to post HTTP event notifications (via webhook) on changes to the API state (for example, statuses, events, configuration updates). These events are sent to the specified server URL. You can choose between events being posted in either XML or JSON format. You can subscribe to changes on multiple API parts by registering up to 15 different feedback expressions. The command for registering is **xCommand HttpFeedback**.

The HttpFeedback Register syntax is as follows:

```
FeedbackSlot: <1..4> ServerUrl(r): <S: 1,
2048> Format: <XML/JSON> Expression: <S: 1,
255> Expression: <S: 1, 255> Expression: <S:
1, 255> Expression: <S: 1, 255> Expression:
<S: 1, 255> Expression: <S: 1, 255>
Expression: <S: 1, 255> Expression: <S: 1,
255> Expression: <S: 1, 255> Expression: <S:
1, 255> Expression: <S: 1, 255> Expression:
<S: 1, 255> Expression: <S: 1, 255>
Expression: <S: 1, 255> Expression: <S: 1,
255>
```

Example 10-22 shows a simple Python POST to register a webhook.

Example 10-22 *Python Script to Set a Webhook to Receive Event Notifications*

```
“”” Webex Devices - Set Webhook “””

import requests
URL = "http://10.10.20.159/put.xml"

PAYLOAD = (
    '<Command>' +
    '  <HttpFeedback>' +
    '    <Register command="True">' +
    '      <FeedbackSlot>1</FeedbackSlot>' +
    '      <ServerUrl>http://127.0.0.1/devasc-webhook</ServerUrl>' +
    '      <Format>JSON</Format>' +
    '      <Expression item="1">/Configuration</Expression>' +
    '      <Expression item="2">/Event/CallDisconnect</Expression>' +
    '      <Expression item="3">/Status/Call</Expression>' +
    '    </Register>' +
    '  </HttpFeedback>' +
    '</Command>'
)

HEADERS = {
    'Content-Type': "application/xml",
    'Cookie':
"SessionId=c6ca2fc23d3f211e0517d4c603fbe4205c77d13dd6913c7bc12eef4
085b7637b,SessionId=c6ca2fc23d3f211e0517d4c603fbe4205c77d13dd6913c7bc12eef40
    85b7637b;
SessionId=c6ca2fc23d3f211e0517d4c603fbe4205c77d13dd6913c7bc12eef4085b7
    637b"
}

RESPONSE = requests.request(“POST”, URL, data=PAYLOAD, headers=HEADERS)
print(RESPONSE.text)
```

Room Analytics People Presence Detector

People Presence shows day-to-day conference room usage analytics and helps you find available meeting rooms based on facial recognition and ultrasonic technology. xAPI allows you to turn the detector on/off via an API. Example 10-23 shows a Python script that sets the People Presence detector on a device to ON.

Example 10-23 *Python Script to Set the People Presence Detector to ON*

```
“”” Webex Devices - Set People Presence detector ON “””

import requests

URL = "http://10.10.20.159/put.xml"

PAYLOAD = (
    '<Configuration>' +
    '  <RoomAnalytics>'
    '    <PeoplePresenceDetector>On</PeoplePresenceDetector>' +
    '  </RoomAnalytics>' +
    '</Configuration>'
)

HEADERS = {
    'Content-Type': "application/xml",
    'Cookie':
"SessionId=c6ca2fc23d3f211e0517d4c603fbe4205c77d13dd6913c7bc12eef4085b
7637b"
    }

RESPONSE = requests.request(“POST”, URL, data=PAYLOAD, headers=HEADERS)

print(RESPONSE.text)
```

Cisco Unified Communications Manager

Cisco Unified Communications Manager (Unified CM) is the powerful call-processing component of the Cisco Unified Communications solution. Unified CM supports the following interface types:

- Provisioning interfaces
- Device monitoring and call control interfaces
- Serviceability interfaces
- Routing rules interface

Administrative XML

The Administrative XML (AXL) API provides a mechanism for inserting, retrieving, updating, and removing data from the Unified CM configuration database using an Extensible Markup Language (XML) Simple Object Access Protocol (SOAP) interface. It allows a programmer to access Unified CM provisioning services using XML and exchange data in XML form. The AXL methods, referred to as *requests*, are performed using a combination of HTTP and SOAP. SOAP is an XML remote procedure call protocol. Users perform requests by sending XML data to the Unified CM publisher server, which then returns the AXL response, which is also a SOAP message. For more information, see https://developer.cisco.com/docs/axl/#!axl-developer-guide/overview.

Examples of Unified CM objects that can be provisioned with AXL include the following:

- Unified CM groups
- Call park DNs

- Call pickup groups
- Calling search spaces
- CTI route points
- Device pools
- Device profiles
- Dial plan tags
- Dial plans
- Digit discard instructions
- Directory numbers
- Gateways (analog, T1, PRI)
- Locations
- MGCP devices
- Phones
- Process nodes
- Process node services
- Regions
- Route filters
- Route groups
- Route lists
- Route partitions
- Service parameters
- Translation patterns
- Users
- Voicemail ports

Cisco AXL Toolkit

You can download the AXL Toolkit from the Unified CM Administration page (https://<CUCM-Address>/plugins/axlsqltoolkit.zip), as shown in Figure 10-13.

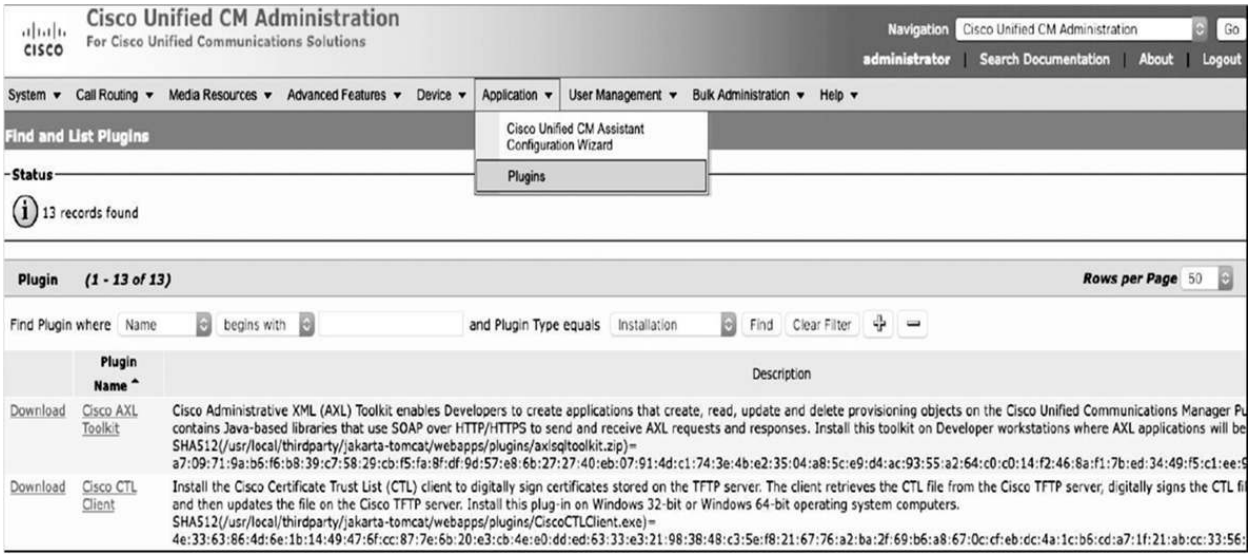


Figure 10-13 *AXL Toolkit Zip File*

When you download the AXL Toolkit and unzip the file, the schema folder contains AXL API schema files for supported AXL versions:

- AXLAPI.wsdl: WSDL file
- **AXLEnums.xsd:** Enum type definitions

- **AXLSoap.xsd:** Type definitions

These XML schema files contain complete details about the AXL API format, including the request names, fields/elements, data types used, and field validation rules. One advantage of .xsd schema files is that they can automatically/programmatically validate a particular XML document against the schema to ensure that it is well-formatted and valid according to the schema specs.

Many XML editing applications and programmatic tools and components can use .xsd schema files to validate XML documents, and details about how to do so may vary. The main necessary detail, however, is to define the **xsi:schemaLocation** attribute with a URI pointing to the axlsoap.xsd schema file.

Accessing the AXL SOAP API

Now, let's look at two different methods to access the AXL SOAP API.

Using the Zeep Client Library

The Python script in Example 10-24 shows how to make an API call to update a phone device and obtain complete information about it. It uses the AXLAPI.wsdl file downloaded with the AXL Toolkit.

Example 10-24 *Full Client Code to Update and Get Back Device Details*

```
“”” Update and Retrieve Client Details “””

from zeep import Client
from zeep.cache import SqliteCache
from zeep.transports import Transport
from requests import Session
from requests.auth import HTTPBasicAuth
import urllib3
from urllib3.exceptions import InsecureRequestWarning

urllib3.disable_warnings(InsecureRequestWarning)

USERNAME = ‘administrator’
PASSWORD = ‘ciscopsdt’
IP_ADDRESS = “10.10.20.1”
WSDL = ‘schema//12.0//AXLAPI.wsdl’
BINDING_NAME = “{http://www.cisco.com/AXLAPIService/}AXLAPIBinding”
ADDRESS = “https://{ip}:8443/axl/”.format(ip=IP_ADDRESS)



def update_phone_by_name(client, name, description):
    """ Update Phone by Name """
    return client.updatePhone(**{'name': name, 'description': description})

def get_phone_by_name(client, name):
    """ Get Phone by Name """
    return client.getPhone(name=name)

def main():
    """ Main """
    session = Session()
    session.verify = False
    session.auth = HTTPBasicAuth(USERNAME, PASSWORD)
    transport = Transport(cache=SqliteCache(), session=session, timeout=60)
    client = Client(wsdl=WSDL, transport=transport)
    axl = client.create_service(BINDING_NAME, ADDRESS)

    update_phone_by_name(axl, "SEP001122334455", "DevAsc: adding new Desc")
    print(get_phone_by_name(axl, "SEP001122334455"))

if __name__ == '__main__':
 main()
```

Using the CiscoAXL SDK

The Python SDK called CiscoAXL is pretty simple to use. As of this writing, other SDKs are available, but we use CiscoAXL here as an example. To start with this SDK, you need to install it by using the **pip** or **pip3** command:

```
pip install ciscoaxl or pip3 install ciscoaxl
```

Example 10-25 shows the simplest way to invoke and start working with the SDK.

Example 10-25 *Using the CiscoAXL SDK to Get Phone Information*

```
"""Using CiscoAXL SDK to get phone info"""


from ciscoaxl import axl


CUCM = '10.10.20.1'
CUCM_USER = "administrator"
CUCM_PASSWORD = "ciscopsdt"
CUCM_VERSION = '12.0'
ucm = axl(username=CUCM_USER,password=CUCM_PASSWORD,cucm=CUCM,cucm_version=CUCM_
VERSION)
print (ucm)
for phone in ucm.get_phones():
     print(phone.name)
for user in ucm.get_users():
     print(user.firstName)
```

For more examples, see Cisco DevNet's Code exchange at https://developer.cisco.com/codeexchange/github/repo/levensailor/ciscoaxl/.

Exam Preparation Tasks

As mentioned in the section "How to Use This Book" in the Introduction, you have a couple of choices for exam preparation: the exercises here, Chapter 19, "Final Preparation," and the exam simulation questions on the companion website.

Review All Key Topics

Review the most important topics in this chapter, noted with the Key Topic icon in the outer margin of the page. Table 10-19 lists these key topics and the page number on which each is found.

Table 10-19 *Key Topics*

Key Topic Element	Description	Page
Paragraph	Webex APIs	
Paragraph	Access Scope in Webex	
Table 10-8	Bot Types	
Example 10-7	Python Code to Generate a JWT Token for a Guest Issuer	
Figure 10-9	Finesse High-Level Flow	
Table 10-13	Webex Meetings Services Supported via APIs	
Paragraph	The Webex XML API	
Paragraph	xAPI	
Paragraph	AXML for Unified CM	

Define Key Terms

Define the following key terms from this chapter and check your answers in the glossary:

Unified Communications
Finesse
JSON Web Token (JWT)
Fully qualified domain name (FQDN)

Chapter 11. Cisco Security Platforms and APIs

This chapter covers the following topics:

- **Cisco Security Portfolio**: This section introduces Cisco's security portfolio and describes how visibility, policy, analytics, and cross-product response fit together in a modern security architecture.
- **Cisco Umbrella:** This section uses Umbrella for the detailed walkthrough because the DNS-layer investigation, enforcement, deployment, and reporting examples map directly to Umbrella APIs, and then compares how Secure Access extends those same patterns into broader SSE and zero-trust workflows.
- **Cisco Firepower and Secure Firewall:** This section provides an overview of the Cisco Firepower and Secure Firewall platform family, Secure Firewall Management Center, and its API categories.
- **Cisco Secure Endpoint** (formerly AMP for Endpoints): This section provides the information you need to understand Cisco's endpoint security platform and the APIs used to investigate and automate response.
- **Cisco Identity Services Engine (ISE):** This section provides an overview of Cisco Identity Services Engine (ISE) and ISE APIs.
- **Cisco Secure Malware Analytics** (formerly Threat Grid): This section provides an overview of Cisco's malware analysis platform and the APIs used to retrieve threat intelligence and sample-analysis data.

Cisco has been building the backbone of the Internet for more than 40 years. In addition, Cisco has created networks, big and small, and gained vast amounts of knowledge about what happens on a network. Along the way, Cisco has built a robust security portfolio to keep networks and users safe. Cisco's security portfolio has three main pillars:

- Visibility
- Segmentation
- Protection

A lot of organizations still struggle with visibility and analytics. It is difficult to protect what you cannot see, and it is even harder to respond quickly when every alert is trapped inside a different tool. That is one reason cross-product security correlation has become so important. Teams need platforms that can help them see activity, understand context, and automate the obvious response steps without forcing analysts to swivel-chair between consoles all day.

This chapter introduces several Cisco security products and some of the APIs used to integrate them into automation workflows. Because Cisco naming has evolved, you will see familiar names such as Firepower, AMP, and Threat Grid alongside the current platform names used today.

- Cisco Firepower
- Cisco Umbrella / Secure Access
- Cisco XDR
- Cisco Secure Endpoint
- Cisco Identity Services Engine (ISE)
- Cisco Secure Malware Analytics

"Do I Know This Already?" Quiz

The "Do I Know This Already?" quiz allows you to assess whether you should read this entire chapter thoroughly or jump to the "Exam Preparation Tasks" section. If you are in doubt about your answers to these questions or your own assessment of your knowledge of the topics, read the entire chapter. Table 11-1 lists the major headings in this chapter and their corresponding "Do I Know This Already?" quiz questions. You can find the answers in Appendix A, "Answers to the 'Do I Know This Already?' Quiz Questions."

Table 11-1 *"Do I Know This Already?" Section-to-Question Mapping*

Foundation Topics Section	Questions
Cisco Security Portfolio	1
Cisco Umbrella	2, 3
Cisco Firepower and Secure Firewall	4, 5
Cisco Secure Endpoint	6, 7
Cisco Identity Services Engine (ISE)	8
Cisco Secure Malware Analytics	9, 10

Caution

The goal of self-assessment is to gauge your mastery of the topics in this chapter. If you do not know the answer to a question or are only partially sure of the answer, you should mark that question as wrong for purposes of self-assessment. Giving yourself credit for an answer that you correctly guess skews your self-assessment results and might provide you with a false sense of security.

1. Cisco's security portfolio focuses on which three areas?
   a. Visibility, segmentation, and protection
   b. Visibility, virtualization, and voice
   c. Segmentation, routing, and switching
   d. Protection, provisioning, and peering
2. The Umbrella _____ API provides enrichment of security events with intelligence to SIEM or other security visibility tool.
   a. Enforcement
   b. Investigate
   c. Management
   d. Reporting
3. The Umbrella Enforcement API involves an HTTP _____ request, which internally comprises an _____ API to check whether the domain is safe.
   a. POST, Reporting
   b. GET, Investigate
   c. POST, Investigate
   d. GET, Reporting
4. In the Firepower Management Center API, the token is generated and returned via _____.
   a. the HTTP header **X-auth-access-token**
   b. JSON object
   c. XML object
   d. JWT
5. A _____ object is a reusable configuration that associates a name with a value.
   a. simple
   b. named
   c. objective
   d. network

6. Which of the following allows bad actors to gain access to and control endpoint resources over an extended period to steal valuable data without being detected?

   a. Malware

   b. Medium persistent threat

   c. Advanced persistent threat

   d. Ransomware

7. Which of the following are capabilities of the Secure Endpoint API? (Choose three.)

   a. Ingesting events into third-party tools

   b. Searching for where a file has been seen or executed

   c. Creating groups and managing file lists

   d. Blocking DNS requests at the resolver

8. Which of the following enables devices and users to be identified and provisioned and enables policies to be applied?

   a. Cisco Secure Endpoint

   b. Cisco ISE

   c. Cisco Secure Malware Analytics

   d. Cisco Firepower Management Center

9. Which of the following describes Cisco Secure Malware Analytics?

   a. A VPN solution from Cisco

   b. A unified malware analysis and threat intelligence platform

   c. A threat intelligence organization

   d. An intelligence security framework

10. Which of the following are used to indicate that a system has been affected by some form of malware?

    a. IoTs

    b. IOCs

    c. Snort

    d. Reports

Foundation Topics

Cisco's Security Portfolio

Cisco's security portfolio is vast. This chapter provides an introduction to some of its critical security products. In addition, it covers the APIs that each of these products supports.

As shown in Figure 11-1, the Cisco security portfolio focuses on three areas: visibility, segmentation, and protection. Visibility is essentially the ability to "see" everything that is happening in the network. Segmentation is the ability of the network to reduce the attack surface to contain the spread. Protection is the ability to help prevent breaches and limit their impact.

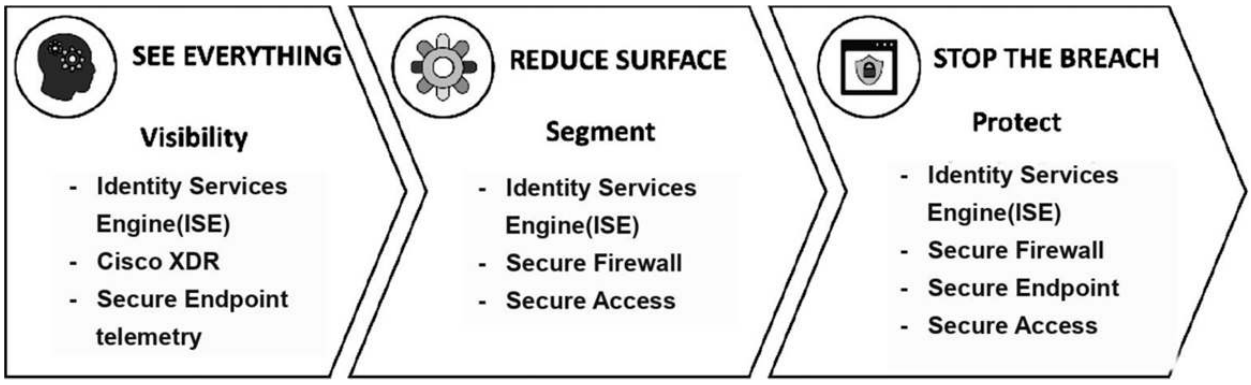


Figure 11-1 *Cisco's Security Portfolio: Visibility, Segmentation, and Protection*

Cisco XDR and Cross-Product Response

Cisco XDR ties these tools together. Rather than treating endpoint, network, DNS, identity, and email telemetry as isolated data sets, XDR correlates them and helps analysts move from observation to action more quickly. Cisco XDR is a cloud-based, API-first platform with OAuth2-protected APIs for automation, investigations, response, enrichment, and playbooks.

- It helps connect findings from multiple Cisco security products and third-party integrations.
- It gives analysts a single place to pivot, investigate, and decide what to do next.
- It makes automation more practical because the same investigation context can feed containment or remediation workflows.

That does not mean every other product disappears. Umbrella still matters for DNS-layer control, Firepower still matters for firewalling and inspection, ISE still matters for identity and policy, and Secure Endpoint still matters on the host. XDR simply gives the team a better way to correlate what those controls are already telling them.

Potential Threats and Vulnerabilities

Any circumstance or event with the potential to adversely affect an asset or operation is known as a threat. A threat may exploit a vulnerability to compromise confidentiality, integrity, or availability.

A vulnerability is a weakness or gap in protection efforts. It can be exploited by threats to gain unauthorized access to an asset.

Cybersecurity threats comprise a wide range of potentially illegal activities on the Internet. At a very high level, certain activities may be classified into two categories:

- **Malicious activities that target networks or devices directly:** Examples include malware, viruses, and denial-of-service attacks.
- **Malicious activities aided by computer networks or devices:** Examples include fraud, identity theft, phishing scams, information warfare, and cyberstalking.

Most Common Threats

Table 11-2 describes some of the most common threats that can be posed to an entity due to lack of security awareness.

Table 11-2 *Common Threats*

Threat	Description
On-path (man-in-the-middle) attack	Attackers insert themselves between two endpoints (such as a browser and a web server) and intercept or modify communications between the two. The attackers can then collect information as well as impersonate either of the two agents. In addition to websites, these attacks can target email communications, DNS lookups, and public Wi-Fi networks.
Denial-of-service (DoS) attack	An attacker sends multiple requests that flood the server or networks with traffic to exhaust resources and bandwidth. As the system continues with degraded performance, the system becomes more and more nonresponsive, and real requests are left unfulfilled. A DoS attack can be coordinated so that multiple devices launch the attack at the same time. This is known as a distributed denial-of-service (DDoS) attack.

Cross-site scripting (XSS)	Cross-site scripting is an exploit in which the attacker attaches code to a legitimate website that executes when the victim loads that website. When the victim loads the affected content, the browser executes the script as trusted-site content. The script may steal cookies or session tokens, alter the page, or perform other actions in the site's context.
Phishing	This type of exploit involves using emails or web pages to procure sensitive information, such as usernames and passwords.
Malware	Malware is malicious software, such as spyware, ransomware, a virus, or a worm. Malware can be delivered through links, attachments, compromised websites, vulnerable software, or other vectors.
Structured Query Language (SQL) injection	SQL injection is a code injection technique used to modify or retrieve data from SQL databases. By inserting specialized SQL statements into an entry field, an attacker can execute commands that read, modify, or delete data and, in some cases, perform administrative operations.
Brute-force attack	Brute-force methods, which involve using trial and error to decode data, can be used to crack passwords and crack encryption keys. Other targets include API keys, SSH logins, and Wi-Fi passwords.

Cisco Umbrella

Cisco Umbrella is a cloud-based secure gateway that helps protect and defend against threats that arise on the Internet. Cisco Umbrella is the first line of defense for users who may be trying to connect to the Internet from anywhere. A device trying to connect to the Internet needs a DNS (Domain Name System) lookup to translate the name of the site or the service to the IP address that it needs to connect to.

Cisco Umbrella processes billions of DNS requests per day, analyzing and learning about various activities and blocking requests to unwanted and malicious destinations before a connection is even established. Cisco Umbrella incorporates the following security services in its offering, regardless of where the user is located:

- It blocks malware, ransomware, and phishing attempts from malicious or fraudulent sites.
- It can be integrated with Cisco Secure Endpoint and other antivirus or investigation engines.
- It maintains content categories and custom-defined allow lists and block lists to comply with any organization policy.

Understanding Umbrella

Cisco Umbrella processes DNS requests received from users or devices on the networks. DNS-layer protection applies before the subsequent connection, regardless of the application protocol that triggered the lookup.

Let's look at a simple flow using the network shown in Figure 11-2. Say that a user wants to access a site and makes a request to the site. This results in a DNS request being sent to Cisco Umbrella. The following steps occur:

Step 1. Umbrella analyzes the DNS request to check whether the domain is malicious or safe.

Step 2. Umbrella checks to see if any of the policies are triggered, such as content filtering policies or blocked domains.

Step 3. If all is well, the IP address is sent to the requesting user. In the case of stanford.edu, Umbrella returns the correct IP address.

Step 4. When a DNS request is sent for domain.xyz, Umbrella checks whether any policies are triggered or whether this is a known malicious domain.

Step 5. Umbrella responds with a "blocked page" message, informing the user that the domain is either malicious or on the blocked list.

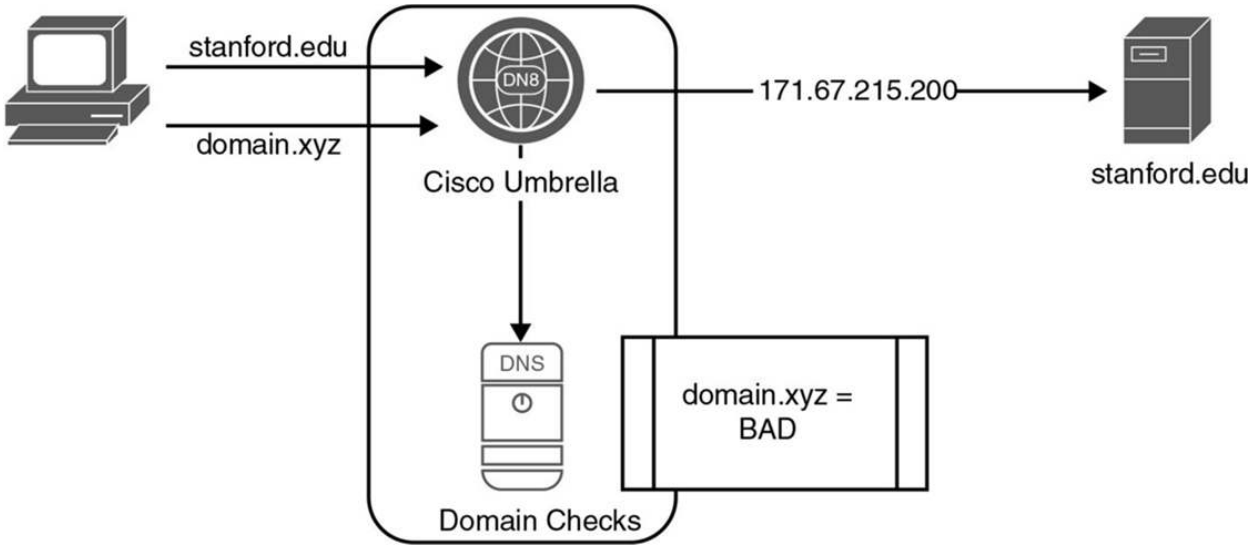


Figure 11-2 *Cisco Umbrella: Blocking a Malicious Domain Lookup*

Cisco Umbrella APIs

Cisco Umbrella exposes several APIs. The key point for learners is that management, investigation, enforcement, and reporting use different interfaces, so the authentication model and endpoint pattern depend on the job you are doing.

This chapter keeps Umbrella as the primary walkthrough because the DNS-layer security examples in this section map directly to Umbrella APIs, especially Investigate, Enforcement, deployment, and reporting workflows. That makes Umbrella the clearest way to learn the API patterns that show up in cloud security automation: bearer-token authentication, family-specific base paths, investigation lookups, policy actions, and reports.

Secure Access builds on that foundation but covers a broader secure service edge use case. Umbrella APIs focus on DNS security, destination control, roaming clients, sites, and Umbrella reporting at api.umbrella.com, whereas Secure Access APIs use api.sse.cisco.com and add broader access, policy, private-resource, tunnel, connector, DLP, and SSE reporting workflows. Both platforms use OAuth 2.0 client credentials with short-lived bearer tokens, and both include investigate, policy, deployment, and reporting API families, so learning Umbrella first makes it easier to recognize how Secure Access extends the same API model. Secure Connect is a separate Meraki-integrated API surface under api.meraki.com/api/secureConnect/v1; its name is not interchangeable with Secure Access.

> **Note**
>
> When you work with Umbrella APIs, pay attention to the specific API family rather than assuming one base URL or version applies everywhere.

Table 11-3 *Umbrella APIs*

API	Description
Management API (legacy)	Legacy v1 organization-management workflows. Cisco identifies the legacy Umbrella APIs as superseded, so treat these older examples primarily as historical context when you encounter existing automation in the field.
Reporting API (legacy v2)	Legacy reporting workflows for request and identity data. Useful for understanding older scripts, but not the place to start when building greenfield automation in 2026.
Console Reporting API (legacy v2)	Legacy reporting for managed-service and multi-organization console views. Current implementations should always be checked against the latest Umbrella developer docs.
Network Device Management API (legacy)	Legacy device-registration workflows for DNS-routing identities. Common in older examples and still helpful when maintaining inherited automation.
Enforcement API	Domain-blocking workflows for sending security events into Umbrella. Review current authentication, rate limits, and lifecycle guidance before using the older sample code verbatim.
Investigate API	Domain and IP enrichment workflows used for classification, correlation, and investigation. This remains one of the most practical Umbrella APIs for security analytics.

Authentication

In the legacy examples in this chapter, management examples use HTTP basic authentication, Investigate examples use bearer-style tokens, and Enforcement examples use the customerKey value passed with the request.

```
"Basic ZGV2YXNjOnN0cm9uZ3Bhc3N3b3Jk"
```

In this case, **ZGV2YXNjOnN0cm9uZ3Bhc3N3b3Jk** is the Base64-encoded string *devasc*:*strongpassword* (where *devasc* is the username, and *strongpassword* is the password). Example 11-1 shows three lines of code that do Base64 encoding to plug the value into the Authorization header.

Example 11-1 *Python Code to Generate Base64 Encoding*

```
""" Generate Base64 encoding using the base64 library """
import base64
encoded =
base64.b64encode('devasc:strongpassword'.encode('UTF-8')).decode('ASCII')
print(encoded)
```

Legacy Management API (v1)

The Umbrella Management API enables direct customers, service providers (SPs), managed service providers (MSPs), and managed security service providers (MSSPs) to manage organizations, networks, network devices, users, and roaming computers and integrate actions in their everyday workflows.

The Management API can use the ISP and MSSP endpoints passing the Base64-encoded authorization in the header (see Example 11-2).

Example 11-2 *Python Code to Get Customer Details Using the Management API*

```
"" Get Customer details given a customerID ""
import requests
url =
"https://management.api.umbrella.com/v1/serviceproviders/serviceProviderId/cu
stomers/customerId"
headers = {
      'accept': "application/json",
      'authorization': "Basic ZGV2YXNjOnN0cm9uZ3Bhc3N3b3Jk"
        }
response = requests.request("GET", url, headers=headers)
print(response.text)
```

The management API includes the following:

- **ISP and MSSP endpoints:** The API returns the service provider IDs and customer IDs. These endpoints are for Internet service providers (ISPs), managed service providers (MSPs) using the Master Service license, managed security service providers (MSSPs), and partner console users. To perform these queries, you must have your service provider ID (SPId) from your console's URL.
- **MSP and multiple-organization endpoints:** The API creates new MSP customers, returns MSP customer records, and updates and deletes MSP customers. This endpoint is for MSP and multiple-organization consoles, *not* for MSPs using MSLA.
- **Networks:** The API returns network records and deletes networks. Note that parent organizations do not have networks. To create and manage networks on behalf of child organizations, use the organizations/customerID/networks endpoints.
- **Roaming computers:** The API returns roaming computer records and updates or deletes roaming computers.
- **Internal networks:** The API creates, updates, and deletes internal networks and returns internal network records.
- **Internal domains:** The API creates, updates, and deletes internal domains and returns internal domain records.
- **Virtual appliances:** The API returns virtual appliance (VA) records and updates or deletes VAs. Note that you cannot create a virtual appliance through the API. A VA must be created within your hypervisor and must be registered as an identity within Umbrella before the API can manage it.
- **Umbrella sites:** The API creates, updates, and deletes sites and returns site records.
- **Users:** The API creates and deletes users and returns user records.
- **Roles:** The API returns a list of roles.
- **Destination lists:** The API creates, reads, updates, and deletes destination lists.

The Enforcement API

The Cisco Umbrella Enforcement API is designed to give technology partners the ability to send security events from their platform/service/appliance within a mutual customer's environment to the Umbrella cloud for enforcement. With this API, you can list domains or delete individual domains from the list.

The API is restricted to HTTPS and is hosted at https://s-platform.api.opendns.com. A fixed UUID-v4 customer key handles customer authentication to the API. A key must be supplied with each request to the API. To generate or get the customer key, you have to log in to the console and navigate to **Policies > Policy Components > Integrations**.

Now let's look at an example of enforcement. Here are the steps involved when the customer detects a malicious domain and wants to add it to Umbrella:

Step 1. The customer identifies malicious code or suspicious activity as users visit a particular URL or domain. The detection can occur with third-party software, Secure Endpoint, or any other mechanism the customer already has in place.

Step 2. This event is sent to the Umbrella Enforcement API via a POST request (see Example 11-3).

Step 3. Cisco Umbrella follows the appropriate logic and algorithm before it adds the domain to the blocked list. It goes through these steps:

a. Umbrella checks whether the domain exists in the Umbrella global block list under one of the security categories.

b. It evaluates the domain with Umbrella reputation and review logic before deciding whether it is benign.

c. It checks on the status of the domain (that is, uncategorized or categorized).

d. It checks to see if the domain is already present on the customer's allow list within the organization.

Step 4. If all the checks are validated, Umbrella blocks domains in that list per that customer's Umbrella policy security settings.

Example 11-3 *Python POST Code to Add a Domain Using the Enforcement API*

```
"""" Add domain using the Enforcement API """"

import json
import requests

url = "https://s-platform.api.opendns.com/1.0/events"

querystring = {"customerKey":"XXXXXXX-YYYY-ZZZZ-YYYY-XXXXXXXXXXXX"}
payload = [
    {
        "alertTime": "2020-01-01T09:33:21.0Z",
        "deviceId": "deadbeaf-e692-4724-ba36-c28132c761de",
        "deviceVersion": "13.7a",
        "dstDomain": "looksfake.com",
        "dstUrl": "http://looksfake.com/badurl",
        "eventTime": "2020-01-01T09:33:21.0Z",
        "protocolVersion": "1.0a",
        "providerName": "Security Platform"
    }
]

headers = {
'Content-Type': "application/json",
    'Accept': "*/*",
    'Cache-Control': "no-cache",
    'Host': "s-platform.api.opendns.com",
    'Accept-Encoding': "gzip, deflate",
    'Connection': "keep-alive",
    'cache-control': "no-cache"
}

response = requests.request(
    "POST",
    url,
data=json.dumps(payload),
    headers=headers,
    params=querystring)

print(response.text)
```

Once domains are placed in the list, a customer can retrieve the current domain list by sending a POST request to the https://s-platform.api.opendns.com/1.0/domains endpoint. Example 11-4 shows a simple Python requests method.

Example 11-4 *Python POST Request to List Domains Using the Enforcement API*

```
"""" List domains using the Enforcement API """"

import requests
url = "https://s-platform.api.opendns.com/1.0/domains"
querystring = {"customerKey":"XXXXXXX-YYYY-ZZZZ-YYYY-XXXXXXXXXXXX"}
response = requests.request("POST", url, headers=headers, params=querystring)
print(response.text)
```

Example 11-5 shows the JSON response to the POST request in Example 11-4. The answer includes a meta tag that describes the page number, the limit, and pointers to the previous and next pages.

Example 11-5 *Response to the POST Request to List Domains Using the Enforcement API*

```
{
    "meta":{
        "page":1,
        "limit":200,
        "prev":false,
        "next":"https://s-platform.api.opendns.com/1.0/domains?customerKey=XX
XXXXX-YYYY-ZZZZ-YYYY-XXXXXXXXXXXX&page=2&limit=200"

 },
    "data":[
        {
    "id":1,
            "name":"baddomain1.com"
        },
        {
            "id":2,
            "name":"looksfake.com"
        },
        {
            "id":3,
            "name":"malware.dom"
        }
    ]
}
```

The final part of the Enforcement API is the **DELETE** call to delete the domain API. The API helps in unblocking domains that were blocked because of previously injected events. Example 11-6 shows an example of a simple Python **requests** command to delete a domain using the Enforcement API. The URL to do this is as follows:

```
https://s-platform.api.opendns.com/1.0/domains/looksfake.
com?customerKey= XXXXXXX-YYYY-ZZZZ-YYYY-XXXXXXXXXXXX
```

Example 11-6 *Deleting Domains Using the Enforcement API*

```
"""" Delete domain using the Enforcement API """"

import requests

url = "https://s-platform.api.opendns.com/1.0/domains/looksfake.com"

querystring = {"customerKey":"XXXXXXX-YYYY-ZZZZ-YYYY-XXXXXXXXXXXX"}

response = requests.request("DELETE", url, headers=headers, params=querystring)
print(response.text)
```

The Legacy Investigate API

The legacy Cisco Umbrella Investigate API is designed to give technology partners the ability to query security events from their platform/service/appliance within a mutual customer's environment to the Umbrella cloud for investigation purposes. The Umbrella Investigate API empowers users to query the Umbrella database to deem a domain safe or not safe. It goes beyond traditional DNS results to show security events and correlations.

The following are some of the tasks that can be done via the Umbrella Investigate REST API:

- Check the security status of a domain, IP address, or subset of domains that appears in the logs from your firewall, UTM, or other Internet egress points.
- Determine whether other related cooccurring domains were accessed at the same time as the domain or IP address you're looking up.
- Find a historical record for this domain or IP address in the DNS database to see what has changed.
- Query large numbers of domains quickly to find out whether they're scored as malicious and require further investigation.

The Investigate API can be accessed via an access token, and the access token can be generated via the Umbrella console.

Developers and customers can use and query Umbrella data via the Investigate API. Here are a few of the standard categories:

- **Categorization (shows the status and classification of the domain):** This category is often used by developers and customers as the primary classifier to determine whether a domain/IP address is good, bad, or unknown.
- **Scoring:** Several scores help rate the potential risk of the domain/IP address. For example:
  - **SecureRank2 (legacy):** This score is designed to identify domains that are requested by known infected clients but never requested by clean clients—assuming that these domains are more likely to be bad. Scores range from –100 (suspicious) to +100 (benign).
  - **RIP Score:** This IP reputation score is designed to rate an IP address based on the amount of malicious activity hosted at that address. Scores range from –100 (very suspicious) to 0.
- **WHOIS record data:** This category includes the email address used to register the domain, the associated name server, historical information, and so on. It can be used to find out more about the history of the domain and the registrant, including whether the email address was used to register other malicious domains.
- **Cooccurrences:** This category depicts other domains that were queried right before or after a given domain and are likely related. The Investigate API is often used to uncover other domains that may be related to the same attack but are hosted on completely separate networks.
- **Passive DNS:** This category depicts the history of domain-to-IP address mappings. This information is used to see if anything suspicious happened with the domain or IP address. For example, you might find that the IP address is continually changing or find that the IP address has more domains than previously declared.
- Malware file data: Information is gathered in the form of malware file analysis and threat intelligence from Secure Malware Analytics. This kind of information is used to find out whether specific malware files are associated with malicious activity seen elsewhere.

Now let's look at a couple of examples of the Investigate API. If you query the domain at https://investigate.umbrella.com/domain-view/name/cisco.com/view via the user interface, you see a screen like the one shown in Figure 11-3.

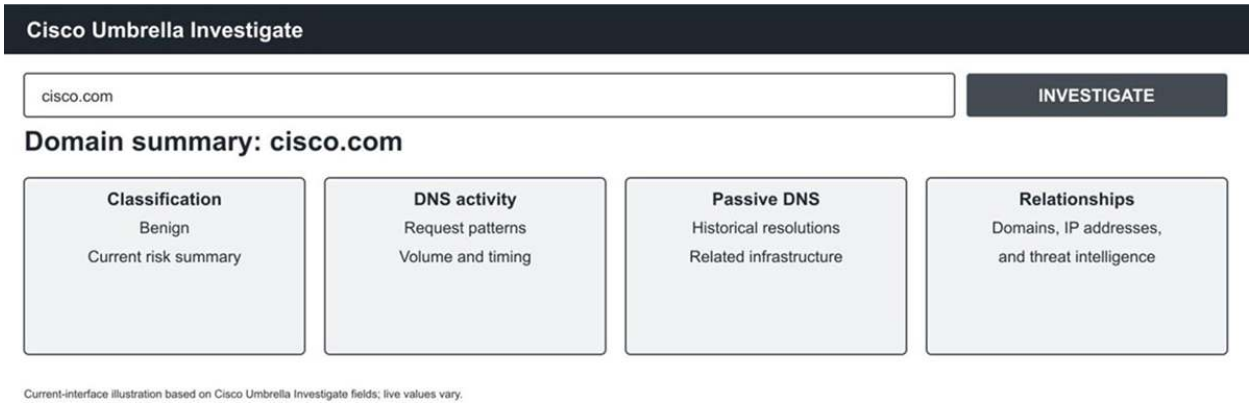


Figure 11-3 *Umbrella Investigate Domain Summary for cisco.com*

In the current interface illustration, cisco.com is classified as benign, and the surrounding panels summarize risk, DNS activity, passive DNS, and related intelligence. Now let's look at the same operation but via the Investigate API. The API to call is https://investigate.api.umbrella.com/domains/categorization/cisco.com. Example 11-7 shows a Python **requests** command that makes an API call with the appropriate API key.

Example 11-7 *Getting Domain Categorization by Using the Investigate API*

```
“”” Domains categorization using the Investigate API “””

import requests
url = “https://investigate.api.umbrella.com/domains/categorization/cisco.com”
querystring = {“showLabels”:””}
headers = {
    ‘authorization’: “Bearer deadbeef-24d7-40e1-a5ce-3b064606166f”,
     ‘cache-control’: “no-cache”,
 }
response = requests.request(“GET”, url, headers=headers, params=querystring)
print(response.text)
```

There are three return values for the categorization:

- **Status:** The status is -1 if the domain is believed to be malicious, 1 if the domain is believed to be benign, or 0 if it hasn't been classified yet.
- **Security:** This field indicates whether there is an Umbrella domain match or whether this domain is associated with one.
- **Content:** This field indicates the type of domain (for example, Ecommerce/Shopping, Business Services).

Example 11-8 shows the response to the request in Example 11-7. Note that the query parameter **showLabels** in Example 11-7 gives the more human-readable information in the response.

Example 11-8 *JSON Returned for Domain Categorization Using the Investigate API*

```
{
    "cisco.com": {
        "status": 1,
        "security_categories": [],
        "content_categories": [
            "Software/Technology",
            "Business Services"
        ]
    }
}
```

Table 11-4 lists other Investigate API URLs for the cisco.com domain.

Table 11-4 *Legacy Umbrella Investigate API Categories*

API Category	API Endpoint URL
Classifiers for a domain	https://investigate.api.umbrella.com/domains/categories/cisco.com.json
Cooccurrences for a domain	https://investigate.api.umbrella.com/recommendations/name/cisco.com.json
Related domains for a domain	https://investigate.api.umbrella.com/links/name/cisco.com.json
Security information for a domain	https://investigate.api.umbrella.com/security/name/cisco.com
Domain volume	https://investigate.api.umbrella.com/domains/volume/cisco.com?start=-2days&stop=now&match=component
Secure Malware Analytics sample for a domain, an IP address, or a URL	https://investigate.api.umbrella.com/samples/cisco.com?limit=100&sortby=score

Cisco Firepower and Secure Firewall

Cisco uses both Firepower and Secure Firewall naming for this product family. In practice, you should recognize Firepower, FMC, FTD, Secure Firewall Management Center, and Secure Firewall Threat Defense as part of the same general platform story.

> **Note**
>
> Current Cisco device output and documentation increasingly use Secure Firewall branding, but engineers still commonly say Firepower, FMC, and FTD in daily work. It helps to be fluent in both sets of names.

Cisco Firepower is Cisco's next-generation firewall (NGFW) platform, combining stateful firewalling with application visibility and control, next-generation intrusion prevention, URL filtering, and advanced malware protection. Firepower also employs other techniques, such as Transport Layer Security/Secure Sockets Layer (TLS/SSL) encrypted traffic inspection, website filtering, quality of service (QoS)/bandwidth management, and malware inspection. It also has built-in management integrations with Lightweight Directory Access Protocol (LDAP), RADIUS, and Active Directory.

In the past, stateful firewalls with simple packet filtering capabilities efficiently blocked unwanted applications because most applications met the port/protocol expectations. NGFWs filter traffic based on the applications or traffic types traversing specific ports. For example, you could open up port 80 for only selected HTTP traffic or for particular applications, sites, or services that you allow. Firepower provides a combination of firewall and QoS functions in a single application-aware solution. Here are the characteristic features of most NGFWs:

- **Standard firewall features:** These include the traditional (first-generation) firewall functionalities such as stateful port/protocol inspection, Network Address Translation (NAT), and virtual private network (VPN) capabilities.
- **Application identification and filtering:** An NGFW identifies and filters traffic based on specific applications rather than just opening ports for all kinds of traffic. An NGFW prevents malicious apps and activities from using nonstandard ports in order to avoid the firewall.
- **SSL and SSH inspection:** NGFWs can inspect SSL- and SSH-encrypted traffic. An NGFW decrypts traffic, makes sure the applications are allowed, checks other policies, and then re-encrypts the traffic. This provides additional protection against malicious apps and activities that try to hide by using encryption to avoid the firewall.

- **Intrusion prevention:** An NGFW has intelligent capabilities to provide more in-depth traffic inspection to perform intrusion detection and prevention.
- **ISE integration:** NGFWs have the support of Cisco ISE. This integration allows authorized users and devices to use specific applications.
- **Malware filtering:** NGFWs can provide reputation-based screening to block applications that have bad reputations. This functionality can check for phishing, viruses, and other malware sites and apps.

Figure 11-4 shows the components of the Firepower solution:

- Firepower Management Center (a management console that has APIs to control and manage application control, URL filtering, malware and file policies, and so on)
- Firepower Threat Defense

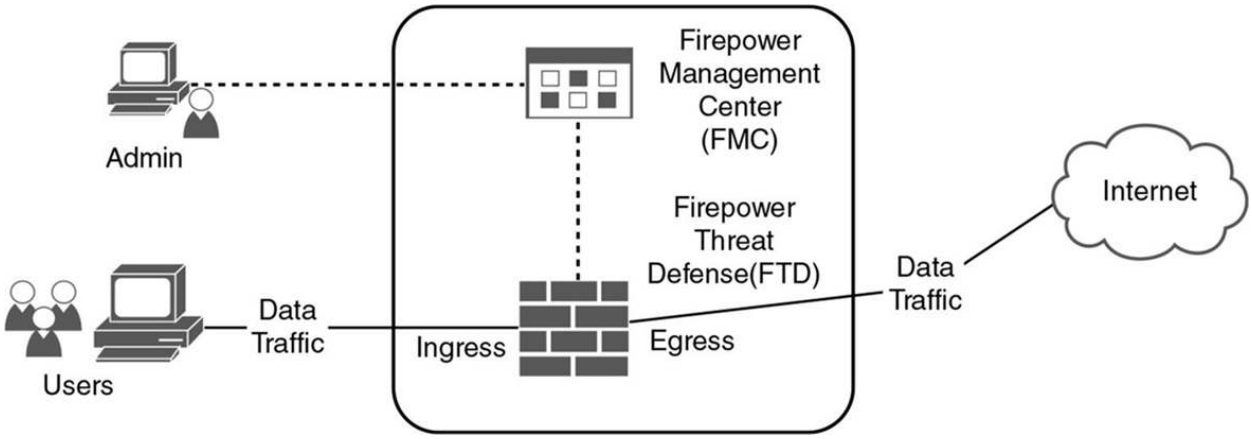


Figure 11-4 *Firepower Components*

Firepower Management Center provides complete and unified management of firewalls, application control, intrusion prevention, URL filtering, and advanced malware protection. It can help administrators smoothly go from managing firewalls to controlling applications to investigating and remediating malware attacks.

Secure Firewall Management Center APIs

Cisco also provides sandbox and reference material for Firepower platforms, including environments that let you explore the Management Center APIs.

The Firepower Management Center REST APIs enable you to program Firepower devices in order to automatically provision devices, discover hosts, perform vulnerability analysis, automate and script firewall configurations, deploy policies and controls, and monitor device health. Figure 11-5 provides an overview of how Firepower Management Center connects to various Firepower devices as well as how you can build your own applications by using REST APIs.

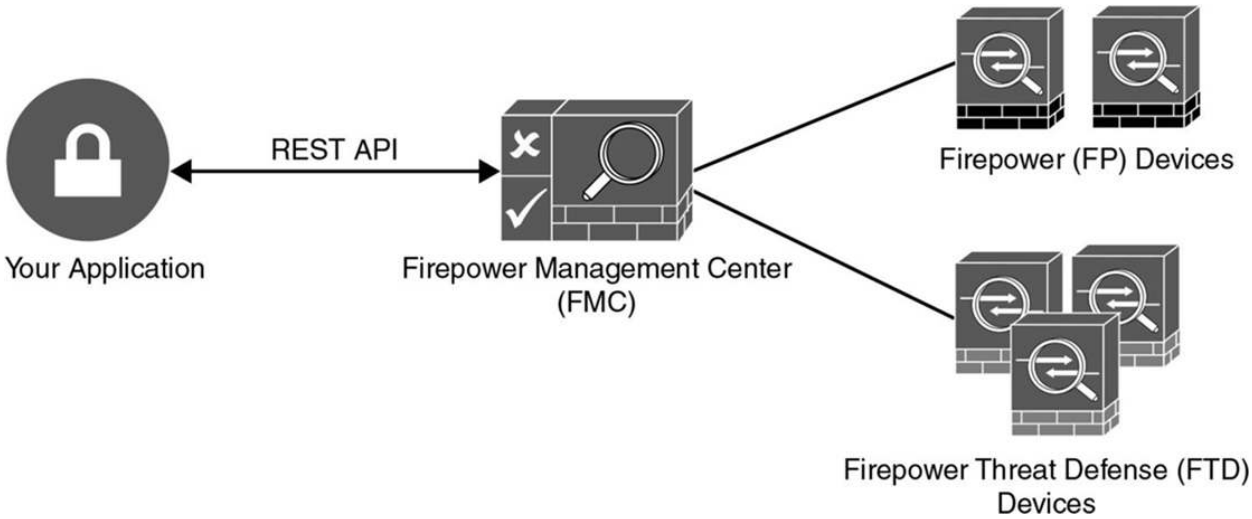


Figure 11-5 *Firepower Allowing Both Northbound and Southbound APIs*

Authentication

The Firepower APIs are already part of the FMC software by default, so the practical task is to verify REST API access in the UI if needed. The Firepower APIs use token-based authentication for API users, and the authentication response provides the access token and related headers that later API calls rely on. Consider the simple Python requests example shown in Example 11-9. The important takeaway is the general token workflow.

Example 11-9 *Python Code to Generate the Session Token for Firepower Management Center*

```
"""  Generate the session token for FMC  """
import requests
import urllib3
urllib3.disable_warnings(urllib3.exceptions.InsecureRequestWarning)

url = "https://<fmc-host>/api/fmc_platform/v1/auth/generatetoken"
headers = {
     'Content-Type': "application/xml",
     'Authorization': "Basic YXNodXRvc2g6V0JVdkE5TXk=",
     }
response = requests.request("POST", url, headers=headers, verify=False)
print(response.headers)
```

The response header contains **'X-auth-access-token': "03d91b3f-eeff-4056- a4a7-e121ddcf8910",** which needs to be used in all subsequent API calls.

System Information

By using **'X-auth-access-token'** you can now make an API call to get the server version. Example 11-10 uses the Python **requests** command to make the REST call, the GET method, and the API https://<fmc-host>/api/fmc_platform/v1/info/serverversion.

Example 11-10 *Python Code to Get the Server Version via the Firepower Management Center API*

```
"""  Get Server Version  """
import requests
url = "https://<fmc-host>/api/fmc_platform/v1/info/serverversion"
headers = {
     'X-auth-access-token': "2abd7bdc-16f8-477f-8022-7f193e71c847",
      }
response = requests.request("GET", url, headers=headers, verify=False)
print(response.text)
```

Now that you know the basics of accessing the API, you can explore all the APIs that Firepower Management Center has to offer.

As indicated earlier, Cisco DevNet provides an instance of Firepower Management Center in the sandbox. The easiest way to figure out specific operations available with any version is by searching for "FMC API Explorer." You can launch the API Explorer by using the URL https://<fmc-host>/api/api-explorer, using the hostname supplied with the sandbox reservation. Figure 11-6 shows the API Explorer, which allows you to explore the version-specific FMC REST API operations and downloadable OAS specifications.

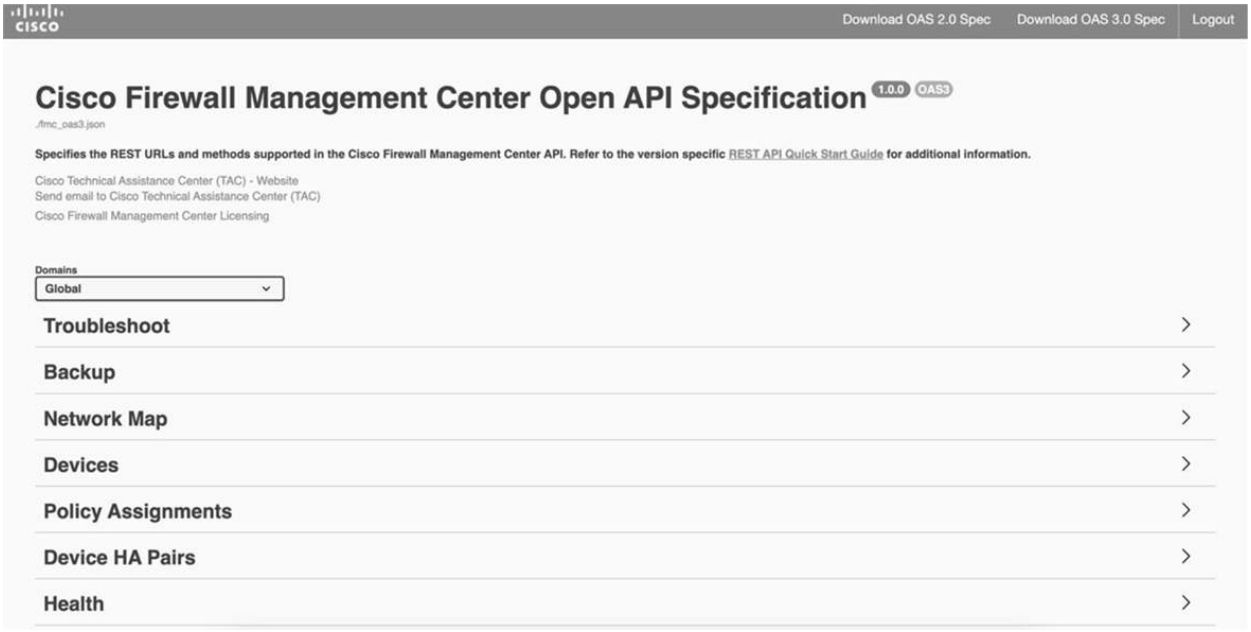


Figure 11-6 *Firewall Management Center Open API Specification (OAS3) Explorer*

Firepower Management Center Objects

In Firepower Management Center, a named object is a reusable configuration that associates a name with a value. When you want to use that value, you can use the named object instead. Firepower Management Center provides many predefined objects that represent frequently used configurations. You can use objects in policies, rules, event searches, reports, and dashboards. The system offers many predefined objects that represent commonly used configurations. Group objects reference multiple objects with a single configuration. Certain predefined objects are groupable, as shown in Table 11-5, which explains the various object types that are defined in Firepower Management Center.

Object Types

Table 11-5 lists the objects you can create in the Firepower system and indicates which object types can be grouped.

Table 11-5 *Firepower Management Center Object Types*

Object Type	Groupable
Network	Yes
Port	Yes
Security zone	No
Application filter	No
VLAN tag	Yes
URL	Yes
Geolocation	No
Variable set	No
Security intelligence: Network, DNS, and URL lists and feeds	No
Sinkhole	No
File list	No
Cipher suite list	No
Distinguished name	Yes
Public key infrastructure (PKI): Internal and trusted CA and internal and external certs	Yes
Route map	No
Community list	No

Creating a Network

A network object represents one or more IP addresses. We use network objects and groups in other objects, including access control policies, network variables, identity rules, network discovery rules, event searches, and reports. Example 11-11 shows how to generate a token, assign it in the header, and then create a network object.

Note that the Firepower Management Center instance in Example 11-11 uses the DevNet Sandbox instance (https://<fmc-host>/) of the FMC, which requires reservation. If you use this example, make sure that you modify the definition of the network object; otherwise, the object creation may fail because the object will be already present. Example 11-11 uses the POST method with the

API /api/fmc_config/v1/domain/" + uuid + "/object/networks.

Example 11-11 *Generating a Session Token and Creating a New Network Object*

```
“”” generate a session token and create a new network object “””

import json
import requests

## Globals used in this file
url = "https://<fmc-host>/api/fmc_platform/v1/auth/generatetoken"
server = "https://<fmc-host>"
username = "johnsmith"
password = "pwgDvQt3"
domain = "Global"
uuid = ""
headers = {
    'Content-Type': "application/json",
}

## Definition of Lab Network (10.10.10.0)

network_lab = {
    "name": "labnetwork-1",
    "value": "10.10.10.0/24",
    "overridable": False,
    "description": "Lab Network Object",
    "type": "Network"
}

def networkObject(network, uuid):
    """ Create a new Network object """

    netpath = "/api/fmc_config/v1/domain/" + uuid + "/object/networks"
    url = server + netpath
    print("--------------------")
    print(headers)
    response = None
    try:
        response = requests.post(url, data=json.dumps(network), headers=headers,
        verify=False)
        status_code = response.status_code
        resp = response.text
        json_response = json.loads(resp)
        print("status code is: " + str(status_code))
        if status_code == 201 or status_code == 202:
            print("Successfully network created")
        else:
            response.raise_for_status()
        return json_response["name"], json_response["id"]
    except requests.exceptions.HTTPError as err:
        print("Reason Code: " + str(err))
    finally:
        if response is not None:
            response.close()

def generateSessionToken():
    """ Generate a new session token using the username and password """
    global uuid
    global headers
    tokenurl = "/api/fmc_platform/v1/auth/generatetoken"
    url = server + tokenurl
    response = requests.request(
        "POST",
        url,
```

```
        headers=headers,
        auth=requests.auth.HTTPBasicAuth(username, password),
        verify=False
    )
    print(response.headers)
    status_code = response.status_code
    if status_code == 204:
        print("Successfully generated session token")
    else:
        response.raise_for_status()

    auth_headers = response.headers
    token = auth_headers.get('X-auth-access-token', default=None)
    headers['X-auth-access-token'] = token
    domains = auth_headers.get('DOMAINS', default=None)
    domains = json.loads("{\"domains\":" + domains + "}")
    for item in domains["domains"]:
        if item["name"] == domain:
            uuid = item["uuid"]
        else:
            print("no UUID for the domain found!")

    print(domains)
    print(uuid)
    print(headers)

## Main - Entry point - Invoke generate token and create network object
if __name__ == "__main__":
    generateSessionToken()
networkObject(network_lab, uuid)
```

Cisco Secure Endpoint

This section provides the information you need to understand Cisco Secure Endpoint. You may also see AMP naming in API hosts, older console labels, and existing examples, so treat those names as referring to the same platform.

Malware is a broad term used to define any malicious activity that aims to infect a network or a specific device. Often, the goal of malware is to steal valuable information, disrupt a user's ability to access data, or cause a device or network to crash. Ransomware is a common form of malware that has become particularly challenging for many companies. A threat actor uses ransomware to encrypt files on an endpoint and extort the owner of the information into paying a ransom to receive the decryption key.

Another common form of malware activity is the advanced persistent threat (APT), where threat actors quietly gain a foothold and then linger long enough to collect data, move laterally, or prepare a later stage of an attack. Secure Endpoint is useful in that kind of fight because it gives defenders visibility into host activity, context around what changed, and controls for taking action without rebuilding every investigative step by hand.

Cisco still hosts some developer material under the older amp-for-endpoints naming, including API hosts and sample repositories. That is normal. The safer way to think about it is this: the platform is Secure Endpoint, while a good portion of the developer surface still carries the older AMP label for compatibility and continuity.

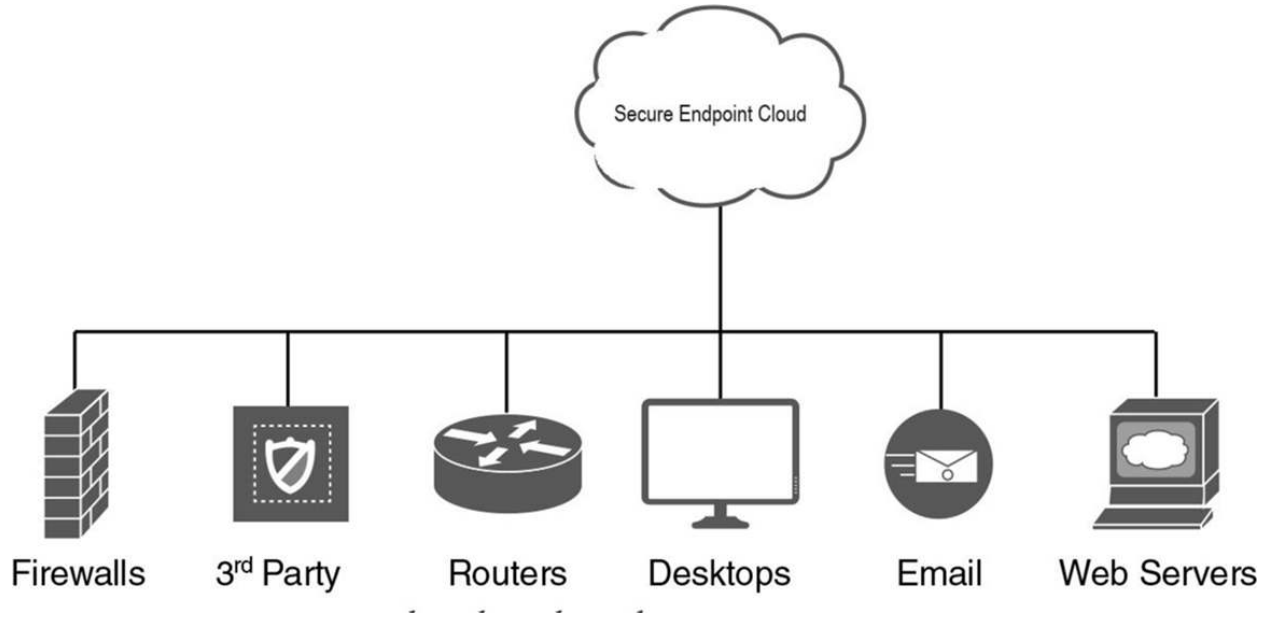


Figure 11-7 *Secure Endpoint Cloud and Endpoints*

With Cisco Secure Endpoint, security teams leverage an integrated solution to proactively and efficiently combat threats in the following ways:

- By streamlining incident response processes: Secure Endpoint allows security operations teams to see where malicious activity appeared, how it moved, and what happened next, which dramatically shortens the time spent hunting blindly for scope.
- By consolidating endpoint security tools into a single solution that makes it possible to visualize the environment, identify vulnerable software, and proactively hunt for malware: Secure Endpoint brings together reputation data, behavior analytics, exploit prevention, and response context in a single management plane.
- By automating threat detection and remediation processes: Customers can reserve analyst time for the ambiguous work instead of spending the whole day on repetitive triage and follow-up tasks.

Automation can be achieved by using the Secure Endpoint API. The older AMP naming still shows up in endpoints and sample code, but the practical goal is the same: identify which endpoints have seen a file, create or manage file lists, move systems in and out of investigative groups, and export event data so it can be correlated with the rest of the security stack.

- **Ingest events:** The API sends events to third-party tools, archives extended event histories, and correlates against other logs.
- **Search:** The API can find where a file has been, determine if a file has been executed, and capture command-line arguments.
- **Basic management:** The API allows you to create groups, move desktops or computers, and manage file lists.

Secure Endpoint API Credentials and Authorization

The legacy examples in this section use the v1 client ID and API-key model. Current v3 workflows register an API client through Cisco XDR or Secure Client Cloud Management and use bearer-token access, but the core tasks remain the same: identify affected endpoints, retrieve security data, and automate response actions.

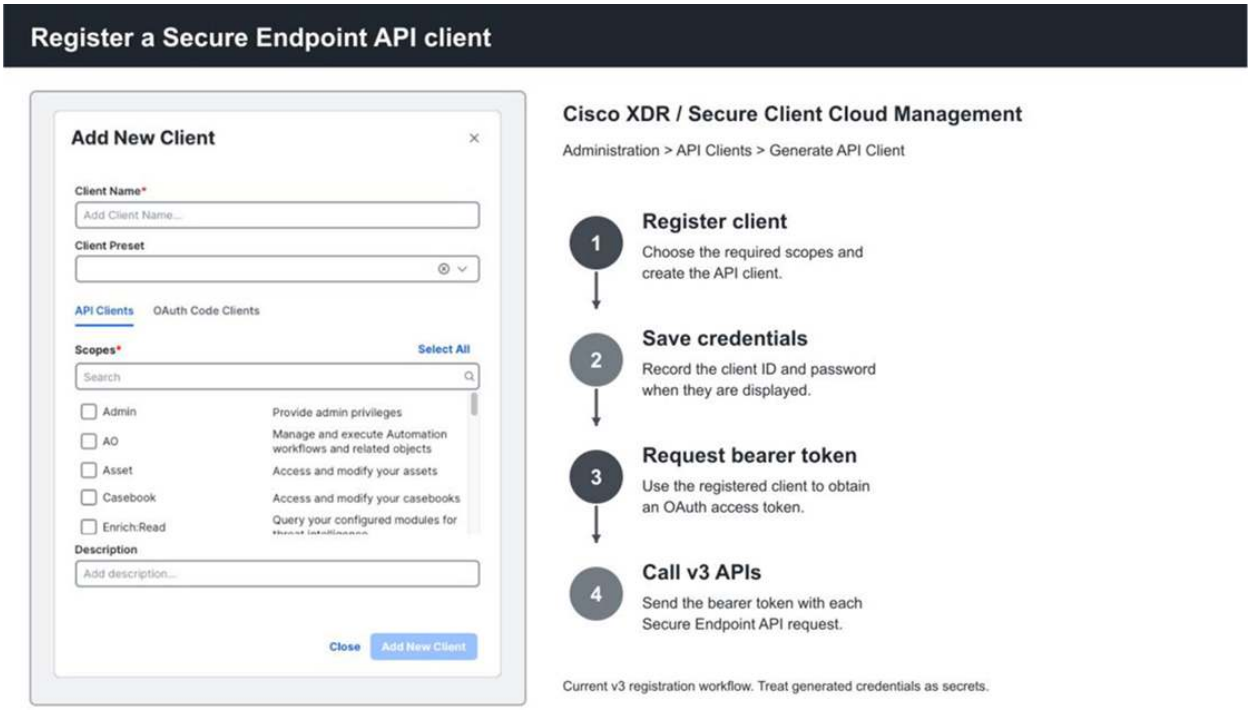


Figure 11-8 *Registering a Secure Endpoint API Client in Cisco XDR*

Once this is done, an API client ID/key pair is generated. It looks something like this:

- **Client ID:** deadbeef123448ccc00d
- **Client key:** XXXXXXXX-YYYY-ZZZZ-0000-e384ef2dxxxx

Using the API client ID and key, you can now make the API calls as follows:

```
https://<clientID>:<clientKEY>@<api_endpoint>
```

Also, you can use basic HTTP authentication encoding for the client ID and key and the Authorization header. For the client ID and key generated, the credential is Base64 encoded as **"ZGVhZGJlZWYxMjM0NDhjY2MwMGQ6WFhYWFhYWFgtWVlZWS1aWlpaLTAwMDAtZTM4NGVmMmR4eHh4",** and the header looks as follows:

```
Authorization: Basic
ZGVhZGJlZWYxMjM0NDhjY2MwMGQ6WFhYWFhYWFgt
WVlZWS1aWlpaLTAwMDAtZTM4NGVmMmR4eHh4
```

Now let's look at a couple of examples of the Secure Endpoint API.

Listing All Computers

The API https://api.amp.cisco.com/v1/computers fetches the list of all computers. It requires basic authentication headers and uses the GET method. Example 11-12 shows a Python **requests** command that uses this API.

Example 11-12 *Python Code to Get a List of All Computers via API*

```
"""  GET list of all computers via API  """
import requests
url = "https://api.amp.cisco.com/v1/computers"
headers = {
    'authorization': "Basic
ZGVhZGJlZWYxMjM0NDhjY2MwMGQ6WFhYWFhYWFgtWVlZWS1aWlpaLTAwMDAtZTM4NGVmMmR4eHh4"
,
     'cache-control': "no-cache",
      }
response = requests.request("GET", url, headers=headers)
print(response.text)
```

Listing All Vulnerabilities

The API https://api.amp.cisco.com/v1/vulnerabilities fetches a list of all vulnerabilities. Treat it as a familiar older example rather than the default starting point for new automation. The list can be filtered to show only the vulnerable programs detected for a specific time range. The start_time and end_time parameters accept the date and time expressed as an epoch value.

The list contains a summary of information such as the following on a vulnerability:

- Application name and version
- SHA-256 value for the executable file
- Connectors on which the vulnerable application was observed
- The most recent CVSS score

This API requires the basic authentication headers and uses the GET method. Example 11-13 shows a Python **requests** command that uses this API.

Example 11-13 *Python Code to Get a List of All Vulnerabilities via API*

```
“”” GET list of all vulnerabilities via API “””
import requests
url = “https://api.amp.cisco.com/v1/vulnerabilities”
querystring = {“offset”:”0”,”limit”:”1”}
headers = {
     ‘authorization’: “Basic
ZGVhZGJlZWYxMjM0NDhjY2MwMGQ6WFhYWFhYWFgtWVlZWS1aWlpaLTAwMDAtZTM4NGVmMmR4eHh4”
,
     ‘cache-control’: “no-cache”,
      }
response = requests.request(“GET”, url, headers=headers, params=querystring)
print(response.text)
```

Example 11-14 shows a sample response to the request in Example 11-13.

Example 11-14 *JSON Response Showing Vulnerabilities*

```
{
  "version": "v1.2.0",
  "metadata": {
    "links": {
      "self": "https://api.amp.cisco.com/v1/vulnerabilities?offset=0&limit=1"
    },
    "results": {
      "total": 1,
      "current_item_count": 1,
      "index": 0,
      "items_per_page": 1
    }
  },
  "data": [
    {
      "application": "Adobe Flash Player",
      "version": "11.5.502.146",
      "file": {
        "filename": "FlashPlayerApp.exe",
        "identity": {
          "sha256": "c1219f0799e60ff48a9705b63c14168684aed911610fec68548ea08f
          605cc42b"
        }
      },
      "cves": [
        {
          "id": "CVE-2013-3333",
          "link":
"https://web.nvd.nist.gov/view/vuln/detail?vulnId=CVE-2013-3333",
          "cvss": 10
        },
        {
          "id": "CVE-2014-0502",
          "link":
"https://web.nvd.nist.gov/view/vuln/detail?vulnId=CVE-2014-0502",
          "cvss": 10
        }
      ],
      "latest_timestamp": 1574442349,
      "latest_date": "2019-11-22T17:05:49+00:00",
      "groups": [
        {
          "name": "Triage",
          "description": "Triage Group for FireAMP API Docs",
          "guid": "68665863-74d5-4bc1-ac7f-5477b2b6406e"
        }
      ],
      "computers_total_count": 1,
      "computers": [
        {
          "connector_guid": "17d71471-805b-4183-9121-3924b8982fac",
          "hostname": "Demo_ZAccess",
          "active": true,
          "links": {
            "computer": "https://api.amp.cisco.com/v1/
            computers/17d71471-805b-4183-9121-3924b8982fac",
            "trajectory": "https://api.amp.cisco.com/v1/
            computers/17d71471-805b-4183-9121-3924b8982fac/trajectory",
            "group": "https://api.amp.cisco.com/v1/
            groups/68665863-74d5-4bc1-ac7f-5477b2b6406e"
          }
```

```
            }
        ]
}
```

Table 11-6 shows other APIs that Secure Endpoint has to offer.

Table 11-6 *Secure Endpoint APIs*

Method	API	Description
GET	https://api.amp.cisco.com/v1/audit_logs	Provide audit logs based on the filters specified in the query parameters
GET	https://api.amp.cisco.com/v1/audit_log_types	Provide a list of all the audit log types supported by the API
GET	https://api.amp.cisco.com/v1/computers	Fetch the list of computers
GET	https://api.amp.cisco.com/v1/computers/user_activity	Fetch the list of computers that have observed activity by a given username
GET	https://api.amp.cisco.com/v1/computers/activity	Search all computers across the organization for any events or activities associated with a file or network operation
GET	https://api.amp.cisco.com/v1/events	Provide a general query interface for events

GET	https://api.amp.cisco.com/v1/event_types	Identify and filter events by a unique ID
POST	https://api.amp.cisco.com/v1/event_streams	Create a new Advanced Messaging Queue Protocol (AMQP) messaging resource for events information
GET	https://api.amp.cisco.com/v1/file_lists/application_blocking	Return a list of application-blocking file lists
GET/POST	https://api.amp.cisco.com/v1/groups	Provide basic information about groups in the organization
GET	https://api.amp.cisco.com/v1/policies	Return a list of policies
GET	https://api.amp.cisco.com/v1/version	Fetch the list of versions
GET	https://api.amp.cisco.com/v1/vulnerabilities	Provide a general query interface for vulnerabilities in the legacy v1 API surface; deprecated for new automation

Cisco Identity Services Engine (ISE)

Cisco Identity Services Engine (ISE) is a network access control and policy enforcement platform. Cisco ISE simplifies the delivery of secure access control across wired and wireless multivendor networks and remote VPN connections. With intelligent sensor and profiling capabilities, ISE penetrates deep to deliver visibility into who and what is accessing your networks and resources.

Cisco ISE provides the following benefits:

- It identifies every device and every user ID across the network.
- It enables simple provisioning for devices.
- It is a simple policy management engine that is centralized and can grant user access.
- It enables flexible integration with other solutions to speed threat detection, containment, and remediation.

Identification is required in order to access any network resources. Identification involves using credentials. Credentials are of the form passwords, certificates, tokens, or at the least the endpoint's MAC address.

Credentials reach Cisco ISE in a process called authentication. An enterprise can use various authentication protocols, depending on the type of network and the type of endpoints. With authentication, you basically tell Cisco ISE who you are.

Authentication typically results in authorization. After you reveal your identity to Cisco ISE, Cisco ISE determines your level of access. The moment an endpoint accesses the network, the network devices generate a session ID and share it with Cisco ISE. Cisco ISE centrally knows what all the endpoints in the network are and where they are connected.

Today, enterprises already have some kind of identity services such as Microsoft Active Directory or LDAP; in addition, there could be other ODBC servers hosting some user and device accounts. A PKI infrastructure may already exist to manage

certificates, and there might be some mobile device managers and identity providers for single sign-on. ISE can seamlessly integrate with all such external identity stores and deliver network access control.

Figure 11-9 shows how Cisco ISE integrates with endpoints, networking devices, and external services.

Once a profile gets associated, various policies can be enforced. These policies could be the following:

- **Time-based:** Policies can allow specific devices only at particular times.
- **Location-based:** Each network element has a piece of location information, and devices connected have specific policies attached.
- **Compliance based:** Policies can ensure that endpoints have all software patches before they are granted full access.

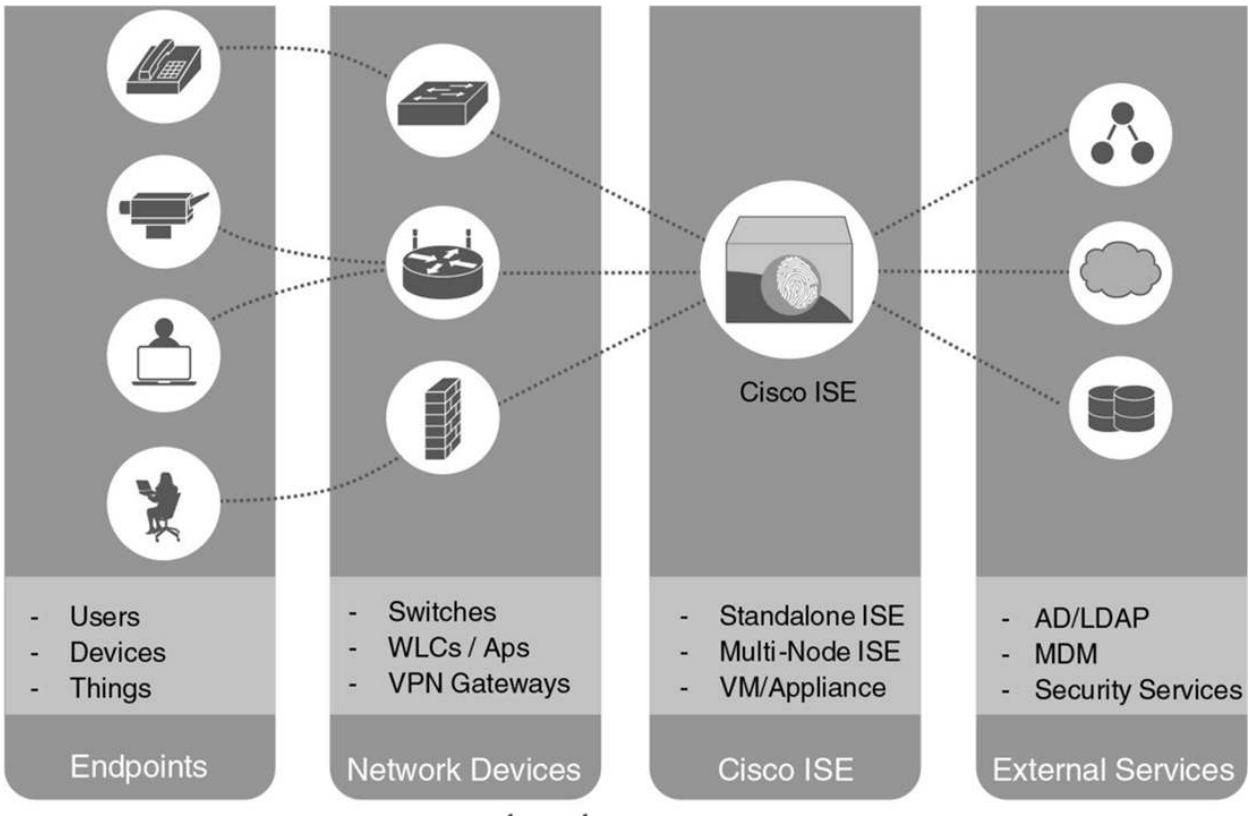


Figure 11-9 *ISE: Components and Deployment*

Cisco ISE APIs

The Cisco DevNet site provides current ISE developer documentation and sandbox resources. The high-level picture is broader than it used to be: ISE still offers practical REST-based automation for common configuration tasks, but the platform also exposes richer integration points for monitoring, context sharing, and data export.

A simple way to think about the current ISE API surface is this:

- Monitoring and session-oriented APIs: These APIs let developers gather session, health, and operational context from ISE so that other tools can see what the policy engine is doing in real time.
- External RESTful Services (ERS) and related Open APIs: These interfaces enable developers to perform configuration and identity operations on ISE resources such as the following:
  - Endpoints
  - Endpoint identity groups
  - Guest users and internal users
  - Identity groups
  - Portals
  - Profiler policies
  - Network devices
  - Network device groups

• Security groups

Cisco ISE also supports pxGrid-based context sharing and Data Connect-style export workflows, which become especially useful when you want ISE identity and posture data to feed a larger analytics or XDR pipeline. The ERS example that follows is still one of the easiest places to start because it shows the mechanics clearly.

> **Note**
>
> The specific API families available to you depend on the ISE version and feature licensing in play, so always verify the exact docs for the release you are automating against.

- **External RESTful Services Admin:** For full access to all ERS methods (GET, POST, DELETE, PUT).
- **External RESTful Services Operator:** For read-only access (GET requests only).

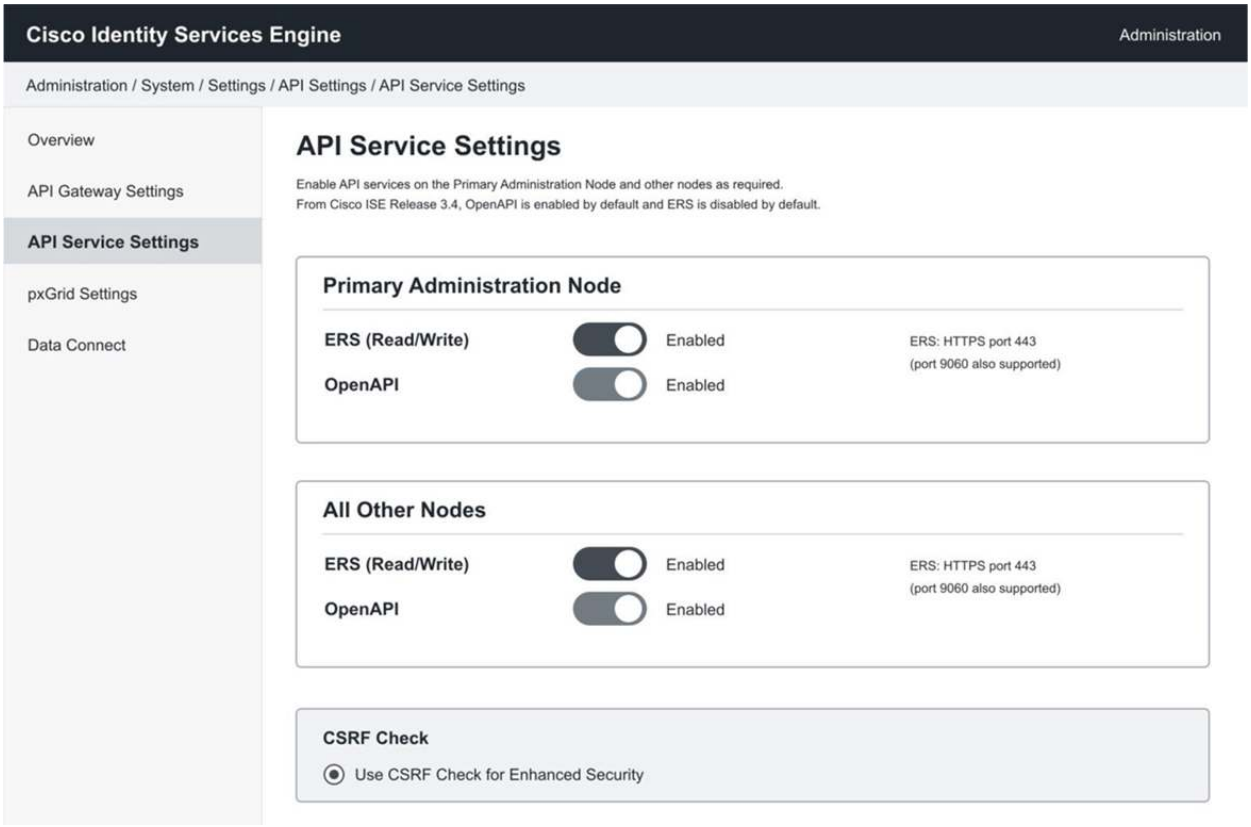


Figure 11-10 *ISE 3.5 API Service Settings for Enabling ERS*

ERS API Authentication

The ISE ERS API uses HTTP basic authentication, which requires the credentials to be sent in the Authorization header. The credentials are the username and password, separated by a colon (**:**), within a Base64-encoded string. For example, the Authorization header would contain the following string:

```
"Basic ZGV2YXNjOnN0cm9uZ3Bhc3N3b3Jk"
```

In this case, **ZGV2YXNjOnN0cm9uZ3Bhc3N3b3Jk** is the Base64-encoded string *devasc*:*strongpassword* (where *devasc* is the username and *strongpassword* is the password). Example 11-15 shows three lines of code that do Base64 encoding to plug the value into the Authorization headers.

Example 11-15 *Python Code to Generate Base64 Encoding*

```
import base64
encoded =
base64.b64encode('devasc:strongpassword'.encode('UTF-8')).decode('ASCII')
print(encoded)
```

ERS APIs are commonly accessed over HTTPS using port 443 or 9060, depending on the ISE release and API gateway configuration.

Now let's look at a couple examples of ISE ERS endpoint APIs.

Creating an Endpoint Group

The API posts the data to create a new endpoint group. It uses the POST method and requires basic authentication headers. The following shows the payload that is needed to create an Endpoint Group called 'DevNet Associate Group':

```
Data - {
      "EndPointGroup" : {
       "name" : "DevNet Associate Group", "description" : "DevNet Associate Group"
     }
}
```

Example 11-16 shows a Python **requests** command using this API.

Example 11-16 *Python POST Code to Create a New Endpoint Group*

```
""" create a new endpointgroup """
import json
import requests
url = "https://<ise-host>:9060/ers/config/endpointgroup"
payload = {
    "EndPointGroup": {
        "name": "DevNet Associate Group",
        "description": "DevNet Associate Group"
    }
}
headers = {
    'content-type': "application/json",
    'accept': "application/json",
'authorization': "Basic ZGV2YXNjOnN0cm9uZ3Bhc3N3b3Jk",
    'cache-control': "no-cache",
}
response = requests.request(
    "POST",
    url,
    data=json.dumps(payload),
    headers=headers
)
print(response.text)
```

The response header contains the newly created group ID:

```
Location:
https://<ise-host>:9060/ers/config/endpointgroup/00000000
-1111-2222-3333-444444444444
```

Creating an Endpoint and Adding It to a Group

The API https://<ise-host>:9060/ers/config/endpoint posts the data to create a new endpoint. It uses the POST method. The following shows the payload that is needed to create an Endpoint called 'DevNet Endpoint' with a specified groupId.

```
Data - {
     "ERSEndPoint" : {
       "name" : "DevNet_Endpoint",
       "description" : "DevNet Endpoint-1",
        "mac" : "FF:EE:DD:03:04:05",
"groupId" : "00000000-1111-2222-3333-444444444444",
         "staticGroupAssignment" : true
      }
}
```

This API uses the group ID from the header and requires basic authentication headers. Example 11-17 shows a Python **requests** script.

Example 11-17 *Python POST Code to Create a New Endpoint*

```
“”” create a new endpoint “””
import json
import requests
url = "https://<ise-host>:9060/ers/config/endpoint"
payload = {
    "ERSEndPoint": {
        "name": "DevNet_Endpoint",
        "description": "DevNet Endpoint-1",
        "mac": "FF:EE:DD:03:04:05",
"groupId": "00000000-1111-2222-3333-444444444444",
        "staticGroupAssignment": True
    }
}
headers = {
    'content-type': "application/json",
    'accept': "application/json",
'authorization': "Basic ZGV2YXNjOnN0cm9uZ3Bhc3N3b3Jk",
    'cache-control': "no-cache",
}
response = requests.request(
    "POST",
    url,
    data=json.dumps(payload),
    headers=headers
)
print(response.text)
```

The response header contains the newly created endpoint ID:

```
Location:
https://<ise-host>:9060/ers/config/endpoint/deadbeef-1111
-2222-3333-444444444444
```

Other ISE APIs

Cisco ISE exposes a broader API surface than the ERS examples shown here, including monitoring, context-sharing, and data-oriented interfaces.

Cisco Secure Malware Analytics

Secure Malware Analytics is Cisco’s unified malware analysis and threat intelligence platform. Earlier Cisco materials call it Threat Grid, and you will still see that older name in API hosts, documentation paths, and sample code. The platform idea is unchanged: upload suspicious artifacts, analyze behavior, and turn the result into intelligence that defenders can actually use.

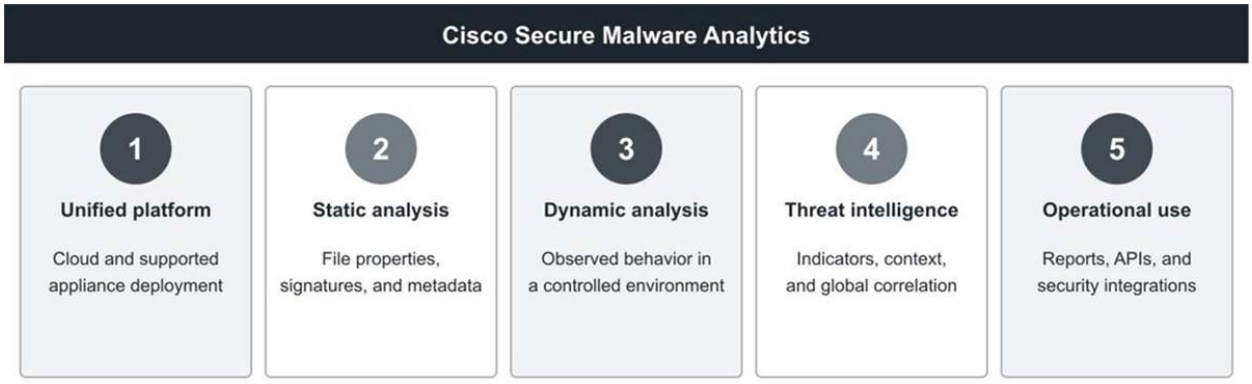


Figure 11-11 *Secure Malware Analytics Capabilities*

Secure Malware Analytics integrates real-time behavioral analysis and current threat intelligence feeds with existing security technologies to help protect a network from both known and unknown attacks. The combination of static analysis, dynamic analysis, and context-rich scoring is what makes the platform especially useful for analysts who need more than a bare malware verdict.

Figure 11-12 shows the Secure Malware Analytics solution architecture and the M6-to-M7 appliance lifecycle.

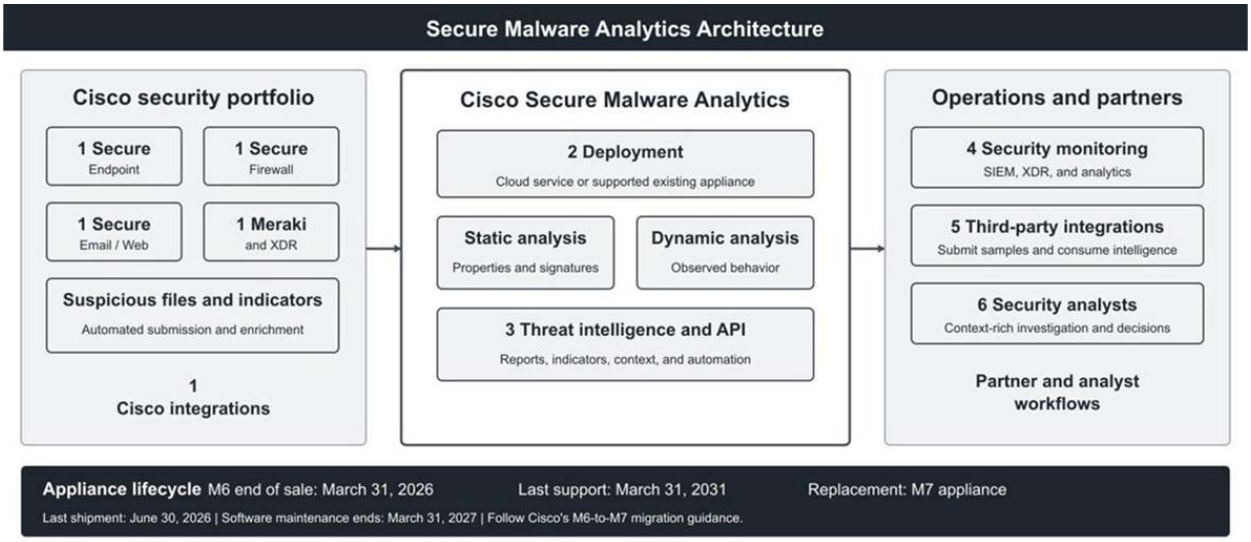


Figure 11-12 *Secure Malware Analytics Architecture and M6-to-M7 Appliance Transition*

On the left side of Figure 11-12 is the Cisco portfolio, and on the right are operations and partner workflows. The numbers in the figure correspond with the following details:

1. The solution can be integrated across the Cisco security portfolio, including Secure Endpoint, Secure Firewall, Secure Email, Secure Web, Meraki, and other partner integrations.
2. Secure Malware Analytics can be deployed as either a cloud-based software-as-a-service product or, for supported existing deployments, an on-premises appliance.
3. A subscription to Secure Malware Analytics provides threat intelligence through the API.
4. Threat intelligence can be delivered to security-monitoring platforms through configured integrations and APIs.
5. Third-party integrations can automatically submit samples and consume threat intelligence when they are configured to do so.
6. Context-rich analysis empowers junior analysts to make more accurate decisions more quickly.

Secure Malware Analytics APIs

The Secure Malware Analytics APIs offer a broad range of functionality, including user and organization account management, sample submission and retrieval, analysis data collection, and threat intelligence harvesting. Here again, the old Threat Grid name may still appear in URLs or examples, so do not mistake that for a different platform.

Secure Malware Analytics API Format

You will still see the Threat Grid host and path format in many working examples. The important lesson is that the request shape varies by endpoint, so focus on the version, endpoint path, and required parameters rather than memorizing one universal template.

A common pattern looks like https://panacea.threatgrid.com/api/<version>/<api-endpoint>, with path and query parameters that vary by endpoint.

In other words, the host and version often stay familiar, but the exact request shape depends on the API you are calling.

API Keys and Access

To get the API key from the Secure Malware Analytics portal UI, follow these steps. In older deployments and screenshots, you may still see the Threat Grid label in the same places.

Step 1. Go to the **Secure Malware Analytics** portal UI.

Step 2. From the **Welcome** menu in the upper-right corner of the navigation bar, select **Manage Users**.

Navigate (use Search if necessary) to the **User Details** page for the integration's user account and copy the API key.

This API key is used in every API call that is made to Secure Malware Analytics, even if the endpoint or script you are maintaining still carries the older Threat Grid name.

The following sections provide some examples of working with the Secure Malware Analytics APIs.

Who Am I

To see if the API key is working, you can use the GET method and the API https://panacea.threatgrid.com/api/v3/session/whoami. You need to pass the API key as a query parameter, as shown in Example 11-18.

Example 11-18 *Secure Malware Analytics: Who Am I*

```
“”” Secure Malware Analytics - who am I “””
import requests
url = “https://panacea.threatgrid.com/api/v3/session/whoami”
querystring = {“api_key”:”deadbeefelcpgib9ec0909”}
headers = {
      ‘cache-control’: “no-cache”,
}
response = requests.request(
“GET”,
url,
headers=headers,
params=querystring
)
print(response.text)
```

Example 11-19 shows the JSON response to the request in Example 11-18.

Example 11-19 *Secure Malware Analytics Response*

```
{
    "api_version": 3,
    "id": 1234567,
    "data": {
        "role": "user",
        "properties": {},
        "integration_id": "z1ci",
       "email": "devasc@student.com",
        "organization_id": 666777,
        "name": "devasc",
        "login": "devasc",
        "title": "DevNet Associate",
        "api_key": " deadbeefelcpgib9ec0909",
        "device": false
    }
}
```

The Data, Sample, and IOC APIs

The Data API allows developers to search observables by specific criteria. You can do an entity search by using the /search/ endpoint and pivot across Secure Malware Analytics data by using entity lookups such as /domains/, /urls/, and /paths/.

The Sample API allows developers to submit and retrieve data for analysis. You can get the raw observable feeds by using the /samples/feeds/ endpoint. The data is usually harvested from all sample activity, suspicious or not, and therefore has a very large footprint. You can use this API to query feeds to look at the sample for your organization only.

The Indicator of Compromise (IOC) API feeds can be accessed via the /iocs/feeds endpoint. With this API, you can see observables in conjunction with behavior indicators. Usually, if an item shows up in this feed, it means that there is at least some degree of suspicious behavior associated with the item. Also, filters can be applied to see only samples from your organization.

Let’s look at an example. In this example, we will search for all records that have a sha1 value equal to **“8fbb3bd96b80e28ea41107728ce8c073cbadb0dd”**. To do so, we use the GET method and the API https://panacea.threatgrid.com/api/v2/search/submissions, and we need to pass the API key as a query parameter. Example 11-20 shows the Python **requests** scripts to use in this case.

Example 11-20 *Secure Malware Analytics: Searching Submissions*

```
“”” Secure Malware Analytics - search submissions API “””
import requests
url = “https://panacea.threatgrid.com/api/v2/search/submissions”
querystring = {
“q”: “8fbb3bd96b80e28ea41107728ce8c073cbadb0dd”,
“api_key”: “deadbeefelcpgib9ec0909”
}
headers = {
      ‘cache-control’: “no-cache”,
}
response = requests.request(
           “GET”,
            url,
            headers=headers,
            params=querystring
)
print(response.text)
```

Example 11-21 shows the JSON response to the request in Example 11-20.

Example 11-21 *Secure Malware Analytics Search Response*

```
{
    "api_version": 2,
    "id": 4482656,
    "data": {
        "index": 0,
        "total": 1,
        "took": 3956,
        "timed_out": false,
        "items_per_page": 100,
        "current_item_count": 1,
        "items": [
            {
                "item": {
                    "properties": {
                        "metadata": null
                    },
                    "tags": [],
                    "vm_runtime": 300,
                    "md5": "b5c26cab57c41208bd6bf92d495ee1f0",
                    "state": "succ",
                    "sha1": "8fbb3bd96b80e28ea41107728ce8c073cbadb0dd",
                    "sample": "b7d3dd2a6d47ea640f719cc76c2122f8",
                    "filename": "FlashPlayer.exe",
....cut
```

Feeds

Secure Malware Analytics supports three different types of feeds:

- **Sample feeds:** These are all observables seen.
- **Indicator of Compromise (IOC) feeds:** These are observables seen in conjunction with behavioral indicators. IOCs are used to indicate that the system has been affected by some form of malware.
- **Curated feeds:** These are highly curated and high-confidence feeds.

Table 11-7 shows the differences between these feeds.

Table 11-7 *Secure Malware Analytics Feeds*

	Sample Feeds	IOC Feeds	Curated Feeds
Version	/v2	/v2	/v3
Endpoint	/samples/feeds/	/iocs/feeds/	/feeds/
Content	All observables are seen	Observables are seen in all BIs	Observables are seen as part of a trusted high-confidence BI triggering
Pre-whitelisted	No	No	Yes
Filterable to only you/org?	Yes	Yes	No
Output Formats	JSON	JSON	JSON/CSV/Snort/ STIX

Say that you want to retrieve all the curated feeds via API. The curated feed types are shown in Table 11-8.

Table 11-8 *Secure Malware Analytics Curated Feed Types*

Feed Name	Description
autorun-registry	Registry entry data derived from querying registry changes known for persistence
banking-dns	Banking Trojan network communications
dga-dns	DGA domains with pseudo-randomly generated names
dll-hijacking-dns	Domains communicated to by samples leveraging DLL sideloading and hijacking techniques
doc-net-com-dns	Document (PDF, Office) network communications
downloaded-pe-dns	Samples downloading executables network communications
dynamic-dns	Samples leveraging dynamic DNS providers
irc-dns	Internet Relay Chat (IRC) network communications
modified-hosts-dns	Modified Windows hosts file network communications
parked-dns	Parked domains resolving to RFC 1918 localhost and broadcast addresses
public-ip-check-dns	Public IP address network communications
ransomware-dns	Samples communicating with ransomware servers
rat-dns	Remote Access Trojan (RAT) network communications
scheduled-tasks	Scheduled task data observed during sample execution
sinkholed-ip-dns	DNS entries for samples communicating with a known DNS sinkhole
stolen-cert-dns	DNS entries observed from samples signed with a stolen certificate

In Example 11-22, we go through all the feed types and printing out the feed if any data exists. In this case, you can use the GET method and the API https://panacea.threatgrid.com/api/v3/feeds. The API key must be passed as a query parameter.

Example 11-22 *Secure Malware Analytics: Listing Details for Each Curated Feed Type*

```
""" Secure Malware Analytics - List details for each curated feed type """
import requests
FEED_URL = "https://panacea.threatgrid.com/api/v3/feeds"
FEEDS_NAME = {
    "autorun-registry": "Autorun Registry Malware",
    "banking-dns": "Banking Trojans",
    "dga-dns": "Domain Generation Algorithm Destinations",
    "dll-hijacking-dns": "DLL Hijackers / Sideloaders",
    "doc-net-com-dns": "Document File Network Communication",
    "downloaded-pe-dns": "Dropper Communication",
    "dynamic-dns": "Dynamic DNS Communication",
    "irc-dns": "IRC Communication",
    "modified-hosts-dns": "Modified HOSTS File Communication",
    "public-ip-check-dns": "Public IP Checkers",
    "ransomware-dns": "Ransomware Communication",
    "rat-dns": "Remote Access Trojans",
    "scheduled-tasks": "Scheduled Task Communication",
    "sinkholed-ip-dns": "Sinkholed IPs",
    "stolen-cert-dns": "Stolen Certificates",
}
for name, desc in FEEDS_NAME.items():
    url = "{}/{}.json".format(FEED_URL, name)
    querystring = {"api_key":"2kdn3muq7uafelcpgib9eccua7"}

    headers = {
        'cache-control': "no-cache",
        'Content-type': 'application/json',
        'Accept': 'application/json'
        }

    response = requests.request("GET", url, headers=headers, params=querystring)

    print(response.text)
```

Exam Preparation Tasks

As mentioned in the section "How to Use This Book" in the Introduction, you have a couple of choices for exam preparation: the exercises here, Chapter 19, "Final Preparation," and the exam simulation questions on the companion website.

Review All Key Topics

Review the most important topics in this chapter, noted with the Key Topic icon in the outer margin of the page. Table 11-9 lists these key topics and the page number on which each is found.

Table 11-9 *Key Topics*

Key Topic Element	Description	Page
Paragraph	Cisco Umbrella	
Paragraph	Enforcement APIs	
Paragraph	Investigate APIs	
Paragraph	Cisco Firepower	
Paragraph	Cisco Secure Endpoint	
Paragraph	Cisco Identity Services Engine (ISE)	
Paragraph	Cisco Secure Malware Analytics	

Define Key Terms

Define the following key terms from this chapter and check your answers in the glossary:

threat
vulnerability
phishing
managed security service provider (MSSP)
next-generation firewall (NGFW)
advanced persistent threat (APT)

Chapter 12. Model-Driven Programmability

This chapter covers the following topics:

- **NETCONF:** This section introduces NETCONF—what it is, why it has been developed, and how to use it.
- **YANG:** This section covers YANG, YANG data models, and how these data models apply to networking.
- **RESTCONF:** This section covers RESTCONF, how it compares to NETCONF, and how to use it.
- **Model-Driven Telemetry:** This section provides a brief introduction to model-driven telemetry.

This chapter introduces the concept of model-driven programmability. It describes what caused data models to be implemented in networking and the advantages they bring to the way networks are configured and monitored. The chapter starts with a short introduction to the network programmability interfaces that network operating systems from Cisco expose. Following that, the chapter provides an in-depth look at NETCONF, why it was necessary to come up with yet another network configuration protocol, and the drawbacks and limitations that this protocol addresses compared to similar protocols. We cannot talk about NETCONF without addressing YANG, so the chapter covers network data modeling next. It ends with RESTCONF and model-driven telemetry. Examples of YANG data models, tools, and libraries to interact with these models, such as pyang and ncclient, are peppered throughout the chapter. The chapter also gives special focus to Cisco IOS-XE and Cisco NX-OS and their implementation of model-driven programmability. Finally, the chapter describes how Postman is used to explore RESTCONF from a hands-on, real-life interaction with the protocol and the data models it supports.

"Do I Know This Already?" Quiz

The "Do I Know This Already?" quiz allows you to assess whether you should read this entire chapter thoroughly or jump to the "Exam Preparation Tasks" section. If you are in doubt about your answers to these questions or your own assessment of your knowledge of the topics, read the entire chapter. Table 12-1 lists the major headings in this chapter and their corresponding "Do I Know This Already?" quiz questions. You can find the answers in Appendix A, "Answers to the 'Do I Know This Already?' Quizzes."

Table 12-1 *"Do I Know This Already?" Section-to-Question Mapping*

Foundation Topics Section	Questions
NETCONF	1–3
YANG	4–6
RESTCONF	7–9
Model-Driven Telemetry	10

Caution

The goal of self-assessment is to gauge your mastery of the topics in this chapter. If you do not know the answer to a question or are only partially sure of the answer, you should mark that question as wrong for purposes of the self-assessment. Giving yourself credit for an answer you correctly guess skews your self-assessment results and might provide you with a false sense of security.

1. What network protocols support model-driven programmability? (Choose three.)
   - **a.** NETCONF
   - **b.** RESTCONF
   - **c.** SSH
   - **d.** gRPC
2. What is the default port on which the NETCONF protocol runs?
   - **a.** TCP 22
   - **b.** TCP 830

c. UDP 161

d. UDP 69

3. What framework does NETCONF use to exchange messages between the client and the server?

a. REST

b. gRPC

c. Apache Avro

d. RPCs

4. Which data type is not a YANG base data type?

a. Binary

b. Enumeration

c. Percent

d. Empty

5. Which component of a YANG module header uniquely identifies a module?

a. Prefix

b. Notification

c. Namespace

d. Organization name

6. Which of the following is a popular Python library used to interact with NETCONF servers?

a. requests

b. ncclient

c. json

d. pyang

7. Which of the following are used for data encapsulation in RESTCONF messages? (Choose two.)

a. XML

b. YANG

c. YAML

d. JSON

8. What NETCONF operation is the equivalent of the RESTCONF PATCH method?

a. <edit-config> (operation="merge")

b. <edit-config> (operation="create/replace")

c. <edit-config> (operation="patch")

d. <patch-config>

9. What RESTCONF resource allows for automatic discovery of the API root?

a. /discover/restconf

b. /.well-known/restconf

c. /.well-known/host-meta

d. /discover/root

10. What types of subscriptions are supported by model-driven telemetry? (Choose two.)

a. Static

b. Dynamic

c. Configured

d. Hard-coded

Foundation Topics

Model-Driven Programmability

Traditionally, network devices have been managed almost exclusively through command-line interfaces (CLIs). For network monitoring, Simple Network Management Protocol (SNMP) is still widely used. While CLIs are extremely powerful, they are also highly proprietary, different from vendor to vendor, and human intervention is required to understand and interpret their output. Also, they do not scale for large network environments. SNMP has its own limitations that will be discussed later in this chapter. A common, standard way of managing and monitoring network devices is needed. Data modeling provides that standards-based, programmatic method of writing configuration data and gathering statistics and operational data from the devices, replacing manual configuration. YANG data models have been developed specifically to address the need for standardization and commonality in network management. Model-driven programmability allows you to automate the configuration and control of network devices.

Cisco IOS-XE (the network operating system for the enterprise), Cisco NX-OS (the network operating system for the data center), and Cisco IOS-XR (the network operating system for service providers) support three standards-based programmable interfaces for operating on the YANG data models: NETCONF, *RESTCONF*, and *gRPC*.

When a client request is received via NETCONF, RESTCONF, or gRPC, it is first converted into an abstract message object. Based on the namespace in the request, the message object is delivered to the underlying model infrastructure, where it is processed and executed on the device data store. The results are returned to the client through the agent from which the request was received.

gRPC is an open-source project started by Google to provide a modern RPC framework that can be used in any environment—not just for network configuration. You can find more details about it at grpc.io. NETCONF and RESTCONF are discussed in detail in this chapter. Figure 12-1 shows all the protocols that Cisco devices support for model-driven programmability.

Figure 12-1 *Model-Driven Programmability on Cisco Devices*

NETCONF

To better address the concerns of network operators, the Internet Engineering Task Force (IETF) and the Internet Architecture Board (IAB) set up a workshop on network management in 2002. Several network operators were invited, and they had a frank discussion about the status of the network management industry as a whole. The conclusions and results of that workshop were captured in RFC 3535. Up to that point, the industry used Simple Network Management Protocol (SNMP) extensively for network management. SNMP will be discussed in more detail later in this book. For the purposes of this discussion, you should know that SNMP is a network management protocol that was initially developed in the late 1980s. The intention with SNMP was to create a protocol to help with configuration and monitoring of network devices. Several iterations of the protocol have been developed through the years; version 3 is the latest one.

Network vendors implemented SNMP extensively in their devices, and it has been used by network operators. By the middle of 2002, after the IETF/IAB workshop took place, the advantages and shortcomings of the protocol were clear for everybody. SNMP had proved to work reasonably well for monitoring devices, especially when small amounts of information needed to be retrieved. For configuration purposes, the protocol had essentially failed for different reasons, such as lack of writable MIBs, security concerns, and scale constraints. Comparing SNMP with other network management options, such as CLI, HTTP, and XML, network operators came up with a list of requirements for the network management technology of the future. Some of these requirements are summarized here:

- This new network management technology should be easy to use.

- There should be a clear distinction between configuration data and operational data.
- It should be possible to configure extensive network services as a whole (such as IPTV, L3VPNs, etc.) instead of just configuring the network device by device.
- Configuration transactions and easy rollback in case of failure should be supported.
- There should be a standard and consistent representation of the network configuration commands between different vendors.
- A distinction between candidate and active configurations should exist, meaning that devices should be able to hold multiple configurations and activate them as needed.

Based on these requirements, the IETF developed the *Network Configuration Protocol (NETCONF)*. This protocol specifies the means of opening a secure management session with a network device; it includes a set of operations to manipulate the configuration data and retrieve the operational state, and it describes a mechanism to send out notifications. NETCONF was initially defined in 2006 in RFC 4741 and updated in 2011 with RFC 6241. While the NETCONF protocol specifies the mechanism to establish the connection and exchange data between a network administrator and the device, it does not define the actual format of the data.

Defining the data format is the role of the *Yet Another Next Generation (YANG)* data modeling language, which was defined by the IETF in 2010 in RFC 6020 specifically to be used with NETCONF. YANG proved to be so successful and powerful that it has evolved to be used as a general-purpose data modeling language in different programming environments. A data model is simply a well-understood and agreed-upon method to describe "something." As an example, let's consider a simple data model for a person. A person can be classified based on gender (male, female, other), height (feet/inches or meters/centimeters), weight (pounds or kilograms), hair color (brown, blond, black, bald, other), eye color (brown, blue, green, other), to name just a few characteristics. Using this generic data model for a person, you can describe an individual in a way that is easy for others to understand. If I tell you Amy is a woman of 5′8″ and 160 pounds with blond hair and blue eyes, you can easily interpret and understand this data model information for a person.

The NETCONF protocol defines transport over SSH and includes operations and configuration data stores that allow the management of network devices. NETCONF authentication options are the same as for any SSH communication. Username, password, and SSL certificates can be used just as SSH is used with CLI commands.

The protocol implements a client/server communication model. The server side is implemented on the network device through a NETCONF agent that acts as a northbound API. Clients connecting to the agent send partial or complete configuration data and receive state and operational data from the device. Any system that implements the protocol can be a client, but some common ones are Cisco NSO and the ncclient Python library.

Figure 12-2 shows the NETCONF architecture stack and the relationship with YANG.

Figure 12-2 *NETCONF*

The main purpose of NETCONF is to transport data payloads between the client and server. The data payloads can be configuration data, operational and status data, and notifications. NETCONF supports notifications, which are similar to SNMP traps.

Messages sent with NETCONF use *remote procedure calls (RPCs)*, a standard framework for clients to send a request to a server to perform an action and return the results. The client or the manager application sends its XML-formatted message to the server, nesting the request within an <rpc> XML element. The server returns results within an <rpc-reply> element.

As previously mentioned, the first version of NETCONF was defined in 2006, but YANG was not defined until 2010. So, what was used as a payload with the first version of NETCONF? Configuration payloads in those days defaulted to CLI commands. This approach was an advantage because new features exposed with new CLI commands were available immediately over NETCONF, but at the same time this approach brought all the drawbacks of CLI to network automation: unstructured data and vendor-specific and proprietary CLI commands. For this reason, YANG was created as a standard way to define common data models for NETCONF.

NETCONF provides a set of operations to manage device configurations and retrieve status information. These operations include retrieving, configuring, copying, and deleting configuration data stores and retrieving operational data. Based on the NETCONF capabilities that are advertised by the device, additional operations can be supported. Table 12-2 shows a common set of NETCONF operations.

Table 12-2 *NETCONF Operations*

Operation	Description
<get>	Retrieve running configuration and device state information
<get-config>	Retrieve all or part of the specified configuration data store
<edit-config>	Load all or part of a configuration to the specified configuration data store
<copy-config>	Replace an entire configuration data store with another
<delete-config>	Delete a configuration data store
<commit>	Copy the candidate data store to the running data store
<lock> / <unlock>	Lock or unlock the entire configuration data store system
<close-session>	Gracefully terminate the NETCONF session
<kill-session>	Forcibly terminate the NETCONF session

The NETCONF <edit-config> operation supports several attributes:

- **merge:** When this attribute is specified, the configuration data is merged with the configuration at the corresponding level in the configuration data store. This is default behavior.
- **replace:** When this attribute is specified, the configuration data replaces any related configuration in the configuration data store. Only the configuration that is present in the config parameter is affected.
- **create:** When this attribute is specified, the configuration data is added to the configuration only if the configuration data does not exist on the device. If the configuration already exists, an <rpc-error> message is returned with an <error-tag> value data-exists.
- **delete:** When this attribute is specified, the configuration data is deleted from the configuration data store.

NETCONF defines the existence of one or more configuration data stores and allows configuration operations on these data stores. A configuration data store is a set of configuration data that is needed to get the device from its default initial state into a desired operational state. The configuration data stores do not contain operational data. Three types of configuration data stores are defined in NETCONF:

- **running:** This data store holds the complete configuration currently active on the network device. Only one running data store can exist on a device, and it is always present. NETCONF protocol operations refer to this data store with the <running> XML element.
- **candidate:** This data store acts as a workplace for creating and manipulating configuration data. A <commit> operation causes the configuration data contained in it to be applied to the running data store.
- **startup:** This data store contains the configuration data that is loaded when the device boots up and comes online. You need an explicit <copy-config> operation from the <running> data store into the <startup> data store to update the startup configuration with the contents of the running configuration.

The NETCONF server advertises the existence of these data stores through capabilities. When a new session is opened with a NETCONF server, the first message that is sent by the server contains a list of all the capabilities that the server supports.

The information that can be retrieved from a NETCONF server is separated into two classes: configuration data and state data or operational data. Configuration data is the set of writable data that starts the system operations, and the operational data is the non-configuration data of the system, also known as the read-only status data (for example, interface traffic statistics, power supply status).

YANG

As mentioned in RFC 6020, YANG is "a data modeling language used to model configuration and state data manipulated by the Network Configuration Protocol (NETCONF), NETCONF remote procedure calls, and NETCONF notifications." The main motivation behind YANG was to provide a standard way to model configuration and operational data so that network devices can be configured and monitored in a standard and common way. While CLI configuration and monitoring of network devices are user friendly, they are not necessarily network automation friendly. Different network vendors have different CLIs with different features and capabilities. Trying to find a standard way to automate the configuration and monitoring of a heterogeneous network with devices from different vendors was almost impossible before NETCONF and YANG data models were created.

YANG is a language used to model data for the NETCONF protocol. A YANG module defines a hierarchy of data that can be used for NETCONF-based operations. This module allows a complete description of all data sent between a NETCONF client and server. YANG models the hierarchical organization of data as a tree in which each node has a name and either a value or a set of child nodes. YANG provides clear and concise descriptions of the nodes, as well as the interaction between those nodes.

YANG strikes a balance between high-level data modeling and low-level bits-on-the-wire encoding. As a reader of a YANG module, you can see the high-level view of the data model while understanding how the data will be encoded in NETCONF operations.

YANG structures data models into modules and submodules. A module can import data from other external modules and include data from submodules. The hierarchy can be augmented, allowing one module to add data nodes to the hierarchy defined in another module.

A YANG module contains three types of statements:

- Module-header statements describe the module and give information about it.
- Revision statements provide information about the history of the module.
- Definition statements are the body of the module where the data model is defined.

A NETCONF server may implement a number of modules, which is the most common implementation for the Cisco devices that support NETCONF, or it might implement only one module that defines all the available data.

YANG is expressed in XML, meaning that an instance of a YANG model is an XML document.

YANG defines a set of built-in types and has a mechanism through which additional types can be defined. There are more than 20 base types to start with, including the ones in Table 12-3.

Table 12-3 *YANG Built-in Data Types*

Data Type	Description
binary	Binary data
bits	Set of bits
boolean	True or false
decimal64	64-bit signed decimal number
empty	No value
enumeration	Enumerated strings
int8/16/32/64	Integer
uint8/16/32/64	Unsigned integer
string	Unicode string

The **typedef** YANG statement can be used to define derived types from base types. In the following example, a new data type called **percent** is created by limiting a 16-bit unsigned integer value to a range from 0 to 100:

```
typedef percent {

   type uint16 {

      range "0 .. 100";

}

Description "Percentage":

}
```

This new type of data can then be used when building the YANG models. It is common practice to have definitions of new types of data contained in a YANG submodule that will then be imported in the main YANG module.

In RFC 6021, the IETF also defined a large number of YANG types that are commonly used in networking. These data types are organized in the inet-yang-types module. The following are some of the data types defined in this RFC:

- ipv4-address
- ipv6-address
- ip-prefix
- domain-name
- uri
- mac-address
- port-number
- ip-version
- phys-address
- timestamp
- date-and-time
- flow-label
- counter32/64
- gauge32/64

When building YANG models, you can use the **import** statement to make all these data types available in the new model:

```
import "ietf-yang-types" {

   prefix yang;

}
```

When you're importing an external YANG module, the **prefix** statement defines the prefix that will be used when accessing definitions inside the imported module. For example, to reference the IPv4 address data type from the newly imported ietf-yang-types module, you can use the following statement:

```
type yang:ipv4-address;
```

YANG defines four types of nodes for data modeling:

- Leaf nodes
- Leaf-list nodes
- Container nodes
- List nodes

The simplest component of a YANG module is the leaf node. A leaf node has one value of a specific type. An example of a leaf node is the following definition of an interface name:

```
leaf intf-name {

   type string;

   description "The name of the interface";

}
```

The instantiation of this leaf node, also known as the wire representation in a NETCONF message, includes only the XML element and its value:

```
<intf-name>GigabitEthernet0/0</intf-name>
```

A leaf-list is a series of leaf nodes of a specific type. The following list of trunk interfaces is an example of a leaf-list:

```
leaf-list trunk-interfaces {

   type string;

   description "List of trunk interfaces";

}
```

The NETCONF XML instantiation of this model on the wire looks like the following:

```
<trunk-interfaces>TenGigabitEthernet0/1</trunk-interfaces>

<trunk-interfaces>TenGigabitEthernet0/2</trunk-interfaces>

<trunk-interfaces>TenGigabitEthernet0/3</trunk-interfaces>

<trunk-interfaces>TenGigabitEthernet0/4</trunk-interfaces>
```

A container is used to group related nodes within a subtree. It has only child nodes and no value. A container can contain any number of child nodes of any type. Example 12-1 shows a container called **statistics** that contains four leaf nodes of different types.

Example 12-1 *YANG Container Example*

```
container statistics {
        description "A collection of interface statistics.";
        leaf in-octets {
                type yang:counter64;
                description "The total number of octets received on the interface.";
        }
        leaf in-errors {
                type yang:counter32;
                description "Number of inbound packets that contained errors.";
        }
        leaf out-octets {
                type yang:counter64;
                description "The total number of octets sent out on the interface.";
        }
        leaf out-errors {
                type yang:counter32;
                description "Number of outbound packets that contained errors.";
        }
}
```

The XML instantiation of this model is as follows:

```
<statistics>

     <in-octets>5983247896</in-octets>

     <in-errors>2578</in-errors>

     <out-octets>678845633</out-octets>

     <out-errors>0</out-errors>

</statistics>
```

The last type of node that is used for data modeling in YANG is the list. A list defines a sequence of list entries. Each entry is a record instance and is uniquely identified by the values of its key leaves. You can think of a list as a table organized around the key leaf with all the leaves as rows in that table. For example, the user list in Example 12-2 can be defined as having three leaves called name, uid, and full-name, with the name leaf being defined as the key leaf.

Example 12-2 *YANG List Example*

```
list user {
        key name;
        leaf name {
                type string;
        }
        leaf uid {
                type uint32;
        }
        leaf full-name {
                type string;
        }
}
```

In this case, an instantiation of this data model on the wire might look like the model shown in Example 12-3.

Example 12-3 *Instantiation of the YANG List*

```
<user>
        <name>john</name>
        <uid>1000</uid>
        <full-name>John Doe</full-name>
</user>
<user>
        <name>jeanne</name>
        <uid>1001</uid>
        <full-name>Jeanne Doe</full-name>
</user>
```

By combining all the nodes discussed so far, you can create a rudimentary YANG data model. Let's call this module **bogus-interfaces**. You can save the data model defined in this module in a file that has to match the module name. In this case, the file that will contain the module has to be called **bogus-interfaces.yang**.

Several components go into building the YANG data model, as you can see in Figure 12-4.

Figure 12-3 *YANG Module Components*

The YANG header contains the namespace for the module. The namespace is used to uniquely identify each module on a system that implements NETCONF. Next, the prefix is used as short notation for the module so that it can be more easily referenced if necessary. The namespace and prefix statements in YANG have the exact same role in XML documents. Remember, YANG is an XML schema definition language. The header also includes identifying information, such as the name of the organization that created the module, contact information, a short description, and revision information. Revision information is particularly of interest because, as the model evolves, the revision information will change. This way, a user can distinguish between different versions of the data model.

A section of imports and includes follows the header. In this section of the module, information contained in additional modules and submodules is included if needed. The **import** statement refers to definitions in another YANG module. It pulls references from the YANG module, but it does not actually pull in the body of the file that's being imported. It's common to have definitions of data types and groupings defined in a module that are imported and used in several other modules. The **include** statement pulls a submodule into a main module. A module does not have to be contained in a single file. It can easily be split into several submodules for maintenance purposes and for easier separation of content. The **include** statement completely pulls the submodule into the main module.

After the **import** and **include** sections of the data model, an optional section of data type definitions follows. Here, any extra data types that are needed to build the module are defined. In the next section, the actual data model is built. All the configuration and operational data declarations are done in this part. An optional section at the end to define custom RPCs and notifications wraps up the module.

You can define a YANG data model by combining all the data nodes discussed earlier in this chapter. Example 12-4 shows this type of YANG data model.

Example 12-4 *YANG Data Model Example*

```
// Contents of "bogus-interfaces.yang"
module bogus-interfaces {
        namespace "http://bogus.example.com/interfaces";
        prefix "bogus";

        import "ietf-yang-types" {
                prefix yang;
        }

        organization "Bogus Inc.";
        contact "john@bogus.example.com";
        description
                "The module for entities implementing the Bogus interfaces .";

        revision 2020-01-06 {
                description "Initial revision.";
        }
        container interfaces {
                leaf intf-name {
                        type string;
                        description "The name of the interface";
                }

                leaf-list trunk-interfaces {
                        type string;
                        description "List of trunk interfaces";
                }

                container statistics {
                        description "A collection of interface statistics.";
                        leaf in-octets {
                                type yang:counter64;
                                description "Total number of octets received on the interface.";
                        }

                        leaf in-errors {
                                type yang:counter32;
                                description "Number of inbound packets that contained errors.";
                        }

                        leaf out-octets {
                                type yang:counter64;
                                description "Total number of octets sent out on the interface.";
                        }

                        leaf out-errors {
                                type yang:counter32;
                                description "Number of outbound packets that contained errors.";
                        }
                }
                list user {
                        key name;
                        leaf name {
                                type string;
                        }
                        leaf uid {
                                type uint32;
                        }
                        leaf full-name {
                                type string;
                        }
```

```
                    }
                }
}
```

The IETF and other standards-setting groups have defined several YANG models. The following are some of the IETF YANG models:

- RFC 7224: *IANA Interface Type YANG Module*
- RFC 7317: *A YANG Data Model for System Management*
- RFC 7407: *A YANG Data Model for SNMP Configuration*
- RFC 8299: *YANG Data Model for L3VPN Service Delivery*
- RFC 8343: *A YANG Data Model for Interface Management*
- RFC 8344: *A YANG Data Model for IP Management*

You can find a complete, up-to-date collection of all YANG modules at https://github.com/YangModels/yang.

Anyone can create YANG data models. In most cases, though, they are created by network equipment vendors, the IETF, and OpenConfig. OpenConfig is a group of network operators, including AT&T, Google, Microsoft, Facebook, and Apple, that was created to define standards intended to make networks more open and programmable. Since OpenConfig is not a formal standards body, the OpenConfig data models change rapidly.

The data models released by the IETF are also called *open data models*. Open data models provide a common interface across multiple platforms, acting like the lowest common denominator for data models. Native data models are specific for each network vendor and each network operating system. Native models are usually not interoperable with other platforms; they closely mirror the structure of the CLI and define extra features that are specific to each vendor.

YANG is an extensible language, allowing extension statements to be defined by standards bodies, vendors, and individuals. The statement syntax allows these extensions to coexist with standard YANG statements in a natural way. YANG allows a module to augment a data model by inserting additional YANG nodes into the model. This capability enables vendors to add vendor-specific parameters to standard data models in an interoperable way. The **augment** statement allows a module to add data to the schema tree. As an example, let's consider the **statistics** container as being part of the **interfaces-statistics** module. The statistics container is shown in Example 12-5.

Example 12-5 *YANG Data Model Augmentation*

```
container statistics {
        description "A collection of interface statistics.";
        leaf in-octets {
                type yang:counter64;
                description "The total number of octets received on the interface.";
        }
        leaf in-errors {
                type yang:counter32;
                description "Number of inbound packets that contained errors.";
        }
        leaf out-octets {
                type yang:counter64;
                description "The total number of octets sent out on the interface.";
        }
        leaf out-errors {
                type yang:counter32;
                description "Number of outbound packets that contained errors.";
        }
}

In order to augment the module, the "augment" statement is used as follows:

        import interface-statistics {
                prefix "intf-stats";
        }
        augment "/intf-stats:statistics" {
                leaf in-unicast-pkts {
                        type yang:counter64;
                        description "Number of unicast packets received on the interface.";
                }
                leaf out-unicast-pkts {
                        type yang:counter64;
                        description "Number of unicast packets sent out the interface.";
                }
        }
```

Notice in Example 12-5 that the **augment** statement takes the XPath for the **statistics** object in the **intf-stats** module as input. Two new leaves are added to the model: **in-unicast-pkts** for inbound unicast data packets on an interface and **out-unicast-pkts** for outbound unicast data packets on the interface.

YANG supports the definition of NETCONF RPCs. This means that in addition to the predefined NETCONF operations (**get**, **get-config**, **edit-config**), you can define additional operations in YANG. This is done through the **rpc** declaration. It's a common occurrence to have several different operating system images stored in the flash memory of networking devices, including the current running image, the previous image, and an image for upgrade. Upgrading software packages and operating system images is a common operational requirement for network administrators. NETCONF does not support this common operational task by default. Example 12-6 shows the YANG definition of a new NETCONF action to activate a specific software image on the device that implements this action.

Example 12-6 *NETCONF RPC Action Definition*

```
rpc activate-software-image {
        input {
                leaf image {
                        type binary;
                }
        }
        output {
                leaf status {
                        type string;
                }
        }
}
```

The RPC definition contains inputs and outputs. In this case, the input leaf specifies the binary of the image that will be activated, and the output leaf contains the status of the RPC action. Keep in mind that the additional RPCs are defined through YANG models and available on the server side on the device. The client or the management system interacting with the server would also have to be able to interpret the output and execute the custom RPC action.

NETCONF supports notification messages, which are similar to SNMP traps. The **notification** statement is used to define NETCONF notifications. It takes only one argument, which is the identifier, followed by a block of data statements that contain the detailed notification information. Example 12-7 shows a simple NETCONF notification for a configuration change.

Example 12-7 *NETCONF Notification Definition*

```
notification config-change {
        description "The configuration change.";
        leaf operator-name {
                type string;
        }

        leaf-list change {
                type instance-identifier;
        }
}
```

The notification in Example 12-7 has two nodes: one for the name of the operator that performed the change and one for the list of changes performed during the configuration session.

Reading a YANG data model straight from an RFC specification can be a daunting task, especially when you're new to data models. **pyang** is a Python program that has been developed to validate and convert YANG models to various more user-friendly and readable formats. You can download pyang from https://github.com/mbj4668/pyang. The following example uses pyang version 2.1.1. For this example, let's use the **ietf-interfaces** YANG model defined in RFC 8343 and downloaded from https://github.com/YangModels/yang. This data model follows the structure discussed in this chapter; Example 12-8 shows what the **ietf-interfaces.yang** file looks like.

Example 12-8 ***ietf-interfaces*** *YANG Data Model Snippet*

```
module ietf-interfaces {
  yang-version 1.1;
  namespace "urn:ietf:params:xml:ns:yang:ietf-interfaces";
  prefix if;

  import ietf-yang-types {
    prefix yang;
  }

  organization
    "IETF NETMOD (Network Modeling) Working Group";

  contact
    "WG Web:   <https://datatracker.ietf.org/wg/netmod/>
     WG List:  <mailto:netmod@ietf.org>

     Editor:   Martin Bjorklund
               <mailto:mbj@tail-f.com>";

  description
    "This module contains a collection of YANG definitions for
     managing network interfaces.

     Copyright (c) 2018 IETF Trust and the persons identified as
     authors of the code.  All rights reserved.

     Redistribution and use in source and binary forms, with or
     without modification, is permitted pursuant to, and subject
     to the license terms contained in, the Simplified BSD License
     set forth in Section 4.c of the IETF Trust's Legal Provisions
     Relating to IETF Documents
     (https://trustee.ietf.org/license-info).

     This version of this YANG module is part of RFC 8343; see
     the RFC itself for full legal notices.";

  revision 2018-02-20 {
    description
      "Updated to support NMDA.";
```

To run this YANG model through pyang and specify the tree format, you use the command **pyang -f tree ietf-interfaces.yang**. This results in the output shown in Example 12-9.

Example 12-9 *pyang Output for* ***ietf-interfaces.yang***

```
module: ietf-interfaces
  +--rw interfaces
  |  +--rw interface* [name]
  |     +--rw name                        string
  |     +--rw description?                string
  |     +--rw type                        identityref
  |     +--rw enabled?                    boolean
  |     +--rw link-up-down-trap-enable?   enumeration {if-mib}?
  |     +--ro admin-status                enumeration {if-mib}?
  |     +--ro oper-status                 enumeration
  |     +--ro last-change?                yang:date-and-time
  |     +--ro if-index                    int32 {if-mib}?
  |     +--ro phys-address?               yang:phys-address
  |     +--ro higher-layer-if*            interface-ref
  |     +--ro lower-layer-if*             interface-ref
  |     +--ro speed?                      yang:gauge64
  |     +--ro statistics
  |        +--ro discontinuity-time    yang:date-and-time
  |        +--ro in-octets?            yang:counter64
  |        +--ro in-unicast-pkts?      yang:counter64
  |        +--ro in-broadcast-pkts?    yang:counter64
  |        +--ro in-multicast-pkts?    yang:counter64
  |        +--ro in-discards?          yang:counter32
  |        +--ro in-errors?            yang:counter32
  |        +--ro in-unknown-protos?    yang:counter32
  |        +--ro out-octets?           yang:counter64
  |        +--ro out-unicast-pkts?     yang:counter64
  |        +--ro out-broadcast-pkts?   yang:counter64
  |        +--ro out-multicast-pkts?   yang:counter64
  |        +--ro out-discards?         yang:counter32
  |        +--ro out-errors?           yang:counter32
  x--ro interfaces-state
     x--ro interface* [name]
        x--ro name               string
        x--ro type               identityref
        x--ro admin-status       enumeration {if-mib}?
        x--ro oper-status        enumeration
        x--ro last-change?       yang:date-and-time
        x--ro if-index           int32 {if-mib}?
        x--ro phys-address?      yang:phys-address
        x--ro higher-layer-if*   interface-state-ref
        x--ro lower-layer-if*    interface-state-ref
        x--ro speed?             yang:gauge64
        x--ro statistics
           x--ro discontinuity-time    yang:date-and-time
           x--ro in-octets?            yang:counter64
           x--ro in-unicast-pkts?      yang:counter64
           x--ro in-broadcast-pkts?    yang:counter64
           x--ro in-multicast-pkts?    yang:counter64
           x--ro in-discards?          yang:counter32
           x--ro in-errors?            yang:counter32
           x--ro in-unknown-protos?    yang:counter32
           x--ro out-octets?           yang:counter64
           x--ro out-unicast-pkts?     yang:counter64
           x--ro out-broadcast-pkts?   yang:counter64
           x--ro out-multicast-pkts?   yang:counter64
           x--ro out-discards?         yang:counter32
           x--ro out-errors?           yang:counter32
```

This example shows a YANG data model that is 1123 lines long; it has been compressed, and all the critical information has been extracted from it. The clear separation between configuration (**rw**, which means read/write) and state (**ro**, which means read-only) data can be observed right away. All the leaves and groupings in the model and their types are also displayed. pyang is extremely versatile and useful for validating the correctness of YANG models and also converting them in several formats: yin, dsdl, tree, jstree, uml, and jsonxsl.

Let's now explore the NETCONF interface that is provided by the Cisco IOS-XE operating system. Cisco IOS-XE is the network operating system for enterprise switching, routing, and wired and wireless access. The following examples use Cisco IOS-XE 16.6.3 running on a Cisco CSR1000V device with NETCONF enabled. To enable NETCONF on the Cisco CSR1000V device running Cisco IOS-XE version 16.6.3, just issue the **netconf-yang** command in the configuration mode of the device. By default, the NETCONF server on the device runs on TCP port 830 and uses the SSH process for transport. A NETCONF session can be established by using an SSH client and specifying port 830 when connecting to the device. When you're establishing a NETCONF session with a device, a complete list of all the capabilities and YANG modules supported by the device is exchanged with the client. An example of this exchange can be observed in Example 12-10.

Example 12-10 *Capabilities Exchanged When Establishing a NETCONF Session*

```
<?xml version="1.0" encoding="UTF-8"?>
<hello xmlns="urn:ietf:params:xml:ns:netconf:base:1.0">
<capabilities>
<capability>urn:ietf:params:netconf:base:1.0</capability>
<capability>urn:ietf:params:netconf:base:1.1</capability>
<capability>urn:ietf:params:netconf:capability:writable-running:1.0</capability>
<capability>urn:ietf:params:netconf:capability:xpath:1.0</capability>
<capability>urn:ietf:params:netconf:capability:validate:1.0</capability>
<capability>urn:ietf:params:netconf:capability:validate:1.1</capability>
<capability>urn:ietf:params:netconf:capability:rollback-on-error:1.0</capability>
<capability>urn:ietf:params:netconf:capability:notification:1.0</capability>
<capability>urn:ietf:params:netconf:capability:interleave:1.0</capability>
<capability>urn:ietf:params:netconf:capability:with-defaults:1.0?basic-mode=explicit&amp;also-
supported=report-all-tagged</capability>
<capability>urn:ietf:params:netconf:capability:yang-library:1.0?revision=2016-06-21&amp;module-set-
id=0de3f66ee656759ede57b7f3b6cd310d</capability>
<capability>http://tail-f.com/ns/netconf/actions/1.0</capability>
<capability>http://tail-f.com/ns/netconf/extensions</capability>
<capability>http://cisco.com/ns/cisco-xe-ietf-ip-deviation?module=cisco-xe-ietf-ip-
deviation&amp;revision=2016-08-10</capability>
<capability>http://cisco.com/ns/cisco-xe-ietf-ipv4-unicast-routing-deviation?module=cisco-xe-ietf-
ipv4-unicast-routing-deviation&amp;revision=2015-09-11</capability>
<capability>http://cisco.com/ns/cisco-xe-ietf-ipv6-unicast-routing-deviation?module=cisco-xe-ietf-
ipv6-unicast-routing-deviation&amp;revision=2015-09-11</capability>
<capability>http://cisco.com/ns/cisco-xe-ietf-ospf-deviation?module=cisco-xe-ietf-ospf-
deviation&amp;revision=2015-09-11</capability>
<capability>http://cisco.com/ns/cisco-xe-ietf-routing-deviation?module=cisco-xe-ietf-routing-
deviation&amp;revision=2016-07-09</capability>
<capability>http://cisco.com/ns/cisco-xe-openconfig-acl-deviation?module=cisco-xe-openconfig-acl-
deviation&amp;revision=2017-08-21</capability>
<capability>http://cisco.com/ns/mpls-static/devs?module=common-mpls-static-devs&amp;revision=2015-09-
11</capability>
<capability>http://cisco.com/ns/nvo/devs?module=nvo-devs&amp;revision=2015-09-11</capability>
<capability>http://cisco.com/ns/yang/Cisco-IOS-XE-aaa?module=Cisco-IOS-XE-aaa&amp;revision=2017-09-
05</capability>
<capability>http://cisco.com/ns/yang/Cisco-IOS-XE-acl?module=Cisco-IOS-XE-acl&amp;revision=2017-08-
01</capability>
<capability>http://cisco.com/ns/yang/Cisco-IOS-XE-acl-oper?module=Cisco-IOS-XE-acl-
oper&amp;revision=2017-02-07</capability>
…
</capabilities>
<session-id>5546</session-id></hello>]]>]]>
```

All the YANG models supported by the device are included in the hello message as well as a session ID. This version of Cisco IOS-XE supports both NETCONF 1.0 and 1.1. It also supports the writable-running capability (which allows you to write directly

to the running configuration of the device), rollback-on-error 1.0 capability, and notification 1.0 capability. These capabilities translate into YANG models defined by the IETF, as you can clearly see in their XPath definition:

```
urn:ietf:params:netconf:
```

The delimiter string]]>]]> at the end of the response signifies the end of the message. Next, the client needs to respond to the hello message with a list of capabilities that it supports at its turn. The simplest response that a client can send is as follows:

```
<?xml version="1.0" encoding="UTF-8"?>

<hello xmlns="urn:ietf:params:xml:ns:netconf:base:1.0">

<capabilities>

  <capability>urn:ietf:params:netconf:base:1.0</capability>

</capabilities>

</hello>]]>]]>
```

There is no reply from the device to the client's initial hello message, which is expected behavior, but an active NETCONF connection to the device is now established. To retrieve the running configuration of the device, the following XML message is sent next:

```
<?xml version="1.0" encoding="UTF-8"?>

<rpc message-id="101" xmlns="urn:ietf:params:xml:ns:netconf: base:1.0">

   <get-config>

       <source>

         <running/>

       </source>

   </get-config>

</rpc>]]>]]>
```

This example shows a NETCONF RPC message with an ID of 101 defined by the base IETF NETCONF 1.0 namespace. The message ID is used to keep track of the response received from the device. The operation within the message is **get-config** with the target data source of the configuration that is stored in the **running** data store. The response contains the running configuration of the device enclosed in an <rpc-reply> XML element and is displayed in Example 12-11.

Example 12-11 *Snippet of the NETCONF **get-config** Operation for the Running Configuration*

```
<?xml version="1.0" encoding="UTF-8"?>
<rpc-reply xmlns="urn:ietf:params:xml:ns:netconf:base:1.0" message-id="101">
   <data>
      <native xmlns="http://cisco.com/ns/yang/Cisco-IOS-XE-native">
         <version>16.6</version>
         <boot-start-marker />
         <boot-end-marker />
         <service>
            <timestamps>
               <debug>
                  <datetime>
                     <msec />
                  </datetime>
               </debug>
               <log>
                  <datetime>
                     <msec />
                  </datetime>
               </log>
            </timestamps>
         </service>
         <platform>
            <console xmlns="http://cisco.com/ns/yang/Cisco-IOS-XE-platform">
               <output>serial</output>
            </console>
         </platform>
         <hostname>csr1kv</hostname>
         <enable>
            <password>
               <secret>cisco</secret>
            </password>
            <secret>
               <type>5</type>
               <secret>$1$A1eZ$Vp95LGeOXX3CfOKO5K5Nf1</secret>
            </secret>
         </enable>
… omitted output
   </data>
</rpc-reply>
```

While interacting with a NETCONF server this way is possible, this approach is cumbersome and error prone. Several publicly available NETCONF tools and clients make it much easier to interact with a NETCONF server. One popular NETCONF client is the Python 3 ncclient library, so let's use ncclient to explore the Cisco NX-OS NETCONF interface.

Cisco NX-OS is the network operating system that powers Cisco Nexus switches in data centers around the world. Built on top of Linux, Cisco NX-OS (also called Open NX-OS) exposes the full power of Linux, which has rich programmable interfaces and a diverse set of automation tools. The following examples use Cisco NX-OS version 9.2.3, ncclient version 0.6.7, and Python 3.7.4. You can enable NETCONF on Cisco NX-OS version 9.2.3 by issuing the **feature netconf** command in configuration mode. Similar to Cisco IOS-XE, the NETCONF server runs by default on TCP port 830 and uses the SSH protocol as transport.

The **manager** module within the ncclient library handles all NETCONF RPC connections between the client and server. Using the **connect** method available with the **manager** module, you can easily establish a connection to a NETCONF server. As input parameters, the **connect** method takes the hostname or the IP address of the NETCONF server, the port on which the server is running, the username and password that are used to connect, and the hostkey_verify parameter. The hostkey_verify parameter specifies whether the script should look for hostkey verification information in the ~/.ssh/known_hosts location on the server on which the script is run. Example 12-12 shows a simple Python 3 script that retrieves the capabilities of the NETCONF server and displays them to the console.

Example 12-12 *Python Script to Retrieve and Print NETCONF Capabilities*

```
#!/usr/bin/env python

from ncclient import manager

nxos_host = "10.10.10.10"
netconf_port = "830"
username = "admin"
password = "password"

# create a get_capabilities() method
def get_capabilities():
    """
    Method that prints NETCONF capabilities of remote device.
    """
    with manager.connect(host = nxos_host,
                         port = netconf_port,
                         username = username,
                         password = password,
                         hostkey_verify = False) as m:

        # print all NETCONF capabilities
        print('\n***NETCONF Capabilities for device {}***\n'.format(nxos_host))
        for capability in m.server_capabilities:
            print(capability)

if __name__ == '__main__':
    get_capabilities()
```

After importing the **manager** module from **ncclient**, the script defines the NETCONF server IP address, port number, username, and password. The **get_capabilities()** function establishes a connection to the NETCONF server, retrieves the server capabilities with **m.server_capabilities()**, iterates with a for loop over each capability, and displays it to the console. Example 12-13 shows the result of running this script on a Cisco Nexus 9000 switch running Cisco NX-OS 9.2.3 with NETCONF enabled.

Example 12-13 *Output of the Python Script from Example 12-12*

```
***NETCONF Capabilities for device 10.10.10.10***

urn:ietf:params:netconf:base:1.0
urn:ietf:params:netconf:base:1.1
urn:ietf:params:netconf:capability:writable-running:1.0
urn:ietf:params:netconf:capability:rollback-on-error:1.0
urn:ietf:params:netconf:capability:candidate:1.0
urn:ietf:params:netconf:capability:validate:1.1
urn:ietf:params:netconf:capability:confirmed-commit:1.1
http://cisco.com/ns/yang/cisco-nx-os-device?revision=2019-02-17&module=Cisco-NX-OS-
device&deviations=Cisco-NX-OS-device-deviations
```

At this writing, by default, Cisco NX-OS implements the base NETCONF capabilities for version 1.0 and 1.1 as well as the YANG models for writable running configuration, rollback-on-error, candidate data store, and others. Additional YANG models are supported and can be installed manually.

To perform changes on the NETCONF server, you can use the **edit_config** method available with the **manager** module. In Example 12-14, a new loopback interface is created, the administrative state is switched to up, and an IP address is configured on the newly created interface.

Example 12-14 *Python Script to Create a New Loopback Interface*

```
#!/usr/bin/env python

from ncclient import manager

nxos_host = "10.10.10.10"
netconf_port = "830"
username = "admin"
password = "password"
loopback_id = "01"
loopback_ip = "1.1.1.1/32"

# create add_loopback() method
def add_loopback():
    """
    Method that adds loopback interface and configures IP address
    """

    add_loop_interface = """<config>
    <System xmlns="http://cisco.com/ns/yang/cisco-nx-os-device">
        <intf-items>
            <lb-items>
                <LbRtdIf-list>
                    <id>lo{id}</id>
                    <adminSt>up</adminSt>
                    <descr>Intf configured via NETCONF</descr>
                </LbRtdIf-list>
            </lb-items>
        </intf-items>
        <ipv4-items>
            <inst-items>
                <dom-items>
                    <Dom-list>
                        <name>default</name>
                        <if-items>
                            <If-list>
                                <id>lo{id}</id>
                                <addr-items>
                                    <Addr-list>
                                        <addr>{ip}</addr>
                                    </Addr-list>
                                </addr-items>
                            </If-list>
                        </if-items>
                    </Dom-list>
                </dom-items>
            </inst-items>
        </ipv4-items>
    </System>
    </config>""".format(id = loopback_id, ip = loopback_ip)

    with manager.connect(host = nxos_host,
                         port = netconf_port,
                         username = username,
                         password = password,
                         hostkey_verify = False) as m:

        # Add loopback interface
        print("\n Adding Loopback {} with IP address {} to device {}...\n".\
            format(loopback_id, loopback_ip, nxos_host))
        netconf_response = m.edit_config(target='running', \
            config=add_loop_interface)
```

```
        # Print the XML response
        print(netconf_response)

if __name__ == '__main__':
    add_loopback()
```

After importing the manager module from the ncclient library, the script specifies the connection details for the NETCONF server next. The loopback interface ID and IP address that will be created are also defined at this point in the script. The **add_loopback()** function contains the XML document that specifies the NETCONF configuration information in the add_loop_interface variable. The YANG model that is used in this case is specified through the XPath value: **xmlns="http://cisco.com/ns/yang/cisco-nx-os-device**. As you may remember from the previous example, this YANG data model is included by default in Cisco NX-OS. The actual path in the tree-like YANG model where the configuration changes will be performed is also included in the XML document. This XML document matches one to one with the YANG data model Cisco-NX-OS-Device. The script uses the **connect** method next to establish a connection to the NETCONF server and then send the XML configuration document to the **running** data store. The response that is received is displayed to the console using the **print** function. The result of successfully running this script will look similar to the following:

```
Adding Loopback 01 with IP address 1.1.1.1/32 to device 10.10.10.10...



<?xml version="1.0" encoding="UTF-8"?>

<rpc-reply message-id="urn:uuid:6de4444b-9193-4b74-837b-e3994d75a319"

   xmlns="urn:ietf:params:xml:ns:netconf:base:1.0">

   <ok/>

</rpc-reply>
```

RESTCONF

According to RFC 8040, RESTCONF is "an HTTP-based protocol that provides a programmatic interface for accessing data defined in YANG, using the data store concepts defined in the Network Configuration Protocol (NETCONF)." Basically, RESTCONF provides a REST-like interface to the NETCONF–YANG interface model.

Although NETCONF is a significant improvement over SNMP, it doesn't provide network interfaces with a good REST API interface. Rather than developing an entirely new protocol and data model, the IETF extended NETCONF into RESTCONF. RESTCONF is not a replacement for NETCONF. RESTCONF provides an API that aligns with other web application APIs to provide an easy entry point for developers.

Like other REST APIs, RESTCONF uses the HTTPS protocol to encapsulate and send messages. Authentication is accomplished using typical HTTP authentication models, such as basic authentication where usernames and passwords are Base64 encoded and transmitted from the client to the server through an Authentication header. Figure 12-4 shows the RESTCONF protocol stack.

Figure 12-4 *RESTCONF Protocol Stack*

REST APIs typically implement CRUD (create, retrieve, update, and delete) operations, leveraging HTTP methods. RESTCONF maps the NETCONF operations into HTTP methods as shown in Table 12-4.

Table 12-4 *RESTCONF HTTP Methods and Their Corresponding NETCONF Operations*

RESTCONF	NETCONF
GET	<get>, <get-config>
POST	<edit-config> (operation="create")
PUT	<edit-config> (operation="create/replace")
PATCH	<edit-config> (operation="merge")
DELETE	<edit-config> (operation="delete")

The available RESTCONF methods and their corresponding NETCONF operations are as follows:

- The HTTP GET method is sent in the RESTCONF request by the client to retrieve data and metadata for a specific resource. It translates into the NETCONF <get> and <get-config> operations. The GET method is supported for all resource types except operation resources.
- The HTTP POST method is used for NETCONF RPCs and to create a data resource. It represents the same semantics as the NETCONF <edit-config> operation with operation="create".
- The PUT method is used to create or replace the contents of the target resource. It's the equivalent of the NETCONF <edit-config> operation with operation="create/replace".
- The PATCH method provides the framework for a resource patching mechanism; it is the equivalent of the NETCONF <edit-config> operation with operation="merge".
- The HTTP DELETE method is used to delete the target resource; it is the equivalent of the NETCONF <edit-config> with operation="delete".

RESTCONF data is encoded with either XML or JSON. Compared with NETCONF, RESTCONF has added support for the JSON encoding. Two media types are defined for RESTCONF:

- **application/yang.api+xml** for XML encoded payloads
- **application/yang.api+json** for JSON encoded data

One aspect true of all REST APIs is the importance of the URI in identifying the data being requested or configured, and RESTCONF is no exception. A unique characteristic of RESTCONF is that it lacks any true API documentation that a developer would use to learn how to use it. Rather, the YANG models themselves are the API documentation.

All RESTCONF URIs follow the following format:

https://<ADDRESS>/<ROOT>/data/<[YANG_MODULE:]CONTAINER>/<LEAF>[?<OPTIONS>]

where

- ADDRESS is the IP address or the hostname and port number where the RESTCONF agent is available.
- ROOT is the main entry point for RESTCONF requests. Before a device connects to a RESTCONF server, the root must be determined. Per the RESTCONF standard, devices implementing the RESTCONF protocol should expose a resource called **/.well-known/host-meta** to enable discovery of root programmatically.
- data is the RESTCONF API resource type for data. The operations resource type is also available for access to RPC operations.
- [YANG_MODULE:]CONTAINER is the base mode container being used.
- LEAF is an individual element from within the container.
- [?<OPTIONS>] are the options that some network devices may support; they are sent as query parameters that impact the returned results. These options are optional and can be omitted. Here are some examples for possible options:

- **depth = unbounded:** If nothing is specified, this option is the default. It specifies the fact that the returned data should follow the nested models to the end. Integer values specifying the depth of the data to be returned are also supported.
- **content = [all, config, nonconfig]:** This query option controls the type of data returned. If nothing is specified, the default value is **all**.
- **fields = expr:** This option limits what leaves are returned in the response.

Consider the **ietf-interfaces** YANG model defined in RFC 8343, partially presented in Figure 12-5. Using the data model details, you can easily construct the URIs for RESTCONF requests. You can automate the processes to generate the code to support RESTCONF APIs and map those API calls into NETCONF because the mapping from a YANG data model to a RESTCONF URI is known and well defined. RESTCONF helps support a common, REST-based programming model for network automation in general.

Figure 12-5 *Mapping Between the YANG Model and RESTCONF URI*

Enabling RESTCONF on Cisco IOS-XE is simple and straightforward. First, RESTCONF runs over HTTPS, so you enable secure HTTP server on the device by issuing **ip http secure-server** in configuration mode. Next, you enable RESTCONF with the **restconf** command in configuration mode. After these two commands are entered and a network administrator account exists on the device, RESTCONF is ready to be accessed. Cisco IOS-XE 16.6.3 and Postman 7.13 are used in the following examples.

Devices that implement the RESTCONF protocol should expose a resource called /.well-known/host-meta to enable the discovery of the REST API root resource programmatically. Using Postman to perform a GET API call on the https://{{host}}:{{port}}/.well-known/host-meta endpoint results in the output shown in Figure 12-6.

Figure 12-6 *Getting the REST API Root Resource*

The **href** attribute in the response contains the path to the REST API root resource: **/restconf**. Exploring the API further and doing a GET operation on https://{{host}}:{{port}}/restconf will expose the top-level resources available in RESTCONF: data and operations (as shown in Figure 12-7).

Figure 12-7 *Top-Level Resources Available in RESTCONF*

To explore further down the API tree and into the YANG model and to retrieve a complete list of all the interfaces, their status, and traffic statistics through the RESTCONF interface, you use a GET call on the https://{{host}}:{{port}}/restconf/data/ietf-interfaces:interfaces-state/ endpoint. The response received from the API should look similar to the one shown in Figure 12-8.

Figure 12-8 *Getting Interface Statistics with RESTCONF*

Model-Driven Telemetry

Timely collection of network statistics data is critical to ensuring the network performs as expected and to foresee and prevent any impending problems that might arise. Technologies like SNMP, syslog, and CLI have been used historically to gather this state information from the network. Using a pull model to gather the data, in which the request for network data originates from the client, does not scale and restricts automation efforts. With this model, the network device sends data only when the client manually requests it. A push model continuously streams data from the network device to the client. Telemetry enables the push model, providing near instantaneous access to operational data. Clients can subscribe to specific data they need by using standard-based YANG data models delivered over NETCONF.

There are two types of telemetry subscriptions:

- **dynamic:** The subscriber sends a request usually via the ietf-yangpush.yang data model. The device approves the request, replies with a subscription-id, and begins streaming telemetry data. Dynamic subscriptions cannot be changed but can be terminated at any time, usually by closing the NETCONF session.
- **configured:** The subscription is configured via CLI, NETCONF, or RESTCONF and is persistent between reboots. Configured subscriptions can be modified and terminated at any point and can be used to stream data to more than one receiver.

The subscription can specify how notifications are sent to the receivers:

- Periodic notifications are sent with a fixed rate defined in the telemetry subscription. This data is ideal for device counters or measures such as CPU utilization or interface statistics because of their always-changing nature.
- On-change notifications are sent only when the data changes. Some examples for these types of notifications are faults, new neighbors being detected, and thresholds being crossed.

Exam Preparation Tasks

As mentioned in the section "How to Use This Book" in the Introduction, you have a couple of choices for exam preparation: the exercises here, Chapter 19, "Final Preparation," and the exam simulation questions on the companion website.

Review All Key Topics

Review the most important topics in this chapter, noted with the Key Topic icon in the outer margin of the page. Table 12-5 lists these key topics and the page number on which each is found.

Table 12-5 *Key Topics*

Key Topic Element	Description	Page Number
Paragraph	Yang data models: NETCONF, RESTCONF, and gRPC	
Paragraph	NETCONF protocol specification	
Figure 12-2	NETCONF	
Paragraph	NETCONF operations to manage device configurations and retrieve status information	
Paragraph	NETCONF configuration data stores	
Paragraph	YANG and NETCONF	
Paragraph	YANG nodes for data modeling	
Figure 12-3	YANG module components	
Paragraph	YANG modules	
Paragraph	pyang	
Paragraph	RESTCONF	
Paragraph	RESTCONF URIs	
Paragraph	RESTCONF root resource discovery	
Paragraph	Telemetry subscription types	

Define Key Terms

Define the following key terms from this chapter and check your answers in the glossary:

Network Configuration Protocol (NETCONF)
Yet Another Next Generation (YANG)
remote procedure call (RPC)
gRPC
RESTCONF

Chapter 13. Deploying Applications

This chapter covers the following topics:

- **Application Deployment Models:** This section discusses the public, private, hybrid, and edge cloud application deployment models.
- **NIST Definition:** This section covers how NIST SP 800-145 defines cloud computing.
- **Application Deployment Options:** This section describes the private, public, hybrid, community, and edge options for deploying applications in the cloud.
- **Application Deployment Methods:** This section discusses bare-metal virtual machines, containers, and serverless deployments.
- **Bare-Metal Application Deployment:** This section describes running an application directly on dedicated server hardware.
- **Virtualized Applications:** This section describes running applications in virtual machines.
- **Cloud-Native Applications:** This section describes cloud-native applications and how they evolved from virtualization.
- **Containerized Applications:** This section describes containers and how they package application components.
- **Serverless:** This section describes serverless deployment and where it fits.
- **DevOps:** This section provides an overview of the DevOps operational model.
- **Docker:** This section discusses Docker containers and the basic operation of Docker.
- **Application Security:** This section covers protecting secrets, encrypting data, and handling data responsibly.
- **Network Services in Application Deployment:** This section covers the firewalls, DNS, load balancers, and reverse proxies that sit in front of a deployed application.
- **Top OWASP Threats:** This section introduces the OWASP Top 10 web application security risks.

In the past, deployment options for getting an application up and running were fairly simple: You bought a server, fed it hundreds of CD-ROMs, and 12 hours later, you had your application ready to start configuring and inputting data. Thanks to the evolution of application architectures and microservice design patterns, new technologies support the new breed of applications. You now have numerous ways to deploy applications, including automated and flexible ways to get your applications up and providing value to your business. This chapter explores some of the core technologies and operating models that are in use today.

"Do I Know This Already?" Quiz

The "Do I Know This Already?" quiz allows you to assess whether you should read this entire chapter thoroughly or jump to the "Exam Preparation Tasks" section. If you are in doubt about your answers to these questions or your own assessment of your knowledge of the topics, read the entire chapter. Table 13-1 lists the major headings in this chapter and their corresponding "Do I Know This Already?" quiz questions. You can find the answers in Appendix A, "Answers to the 'Do I Know This Already?' Quiz Questions."

Table 13-1 *"Do I Know This Already?" Section-to-Question Mapping*

Foundation Topics Section	Questions
Application Deployment Models	1
NIST Definition	16
Application Deployment Options	2
Application Deployment Methods	3
Bare-Metal Application Deployment	4
Virtualized Applications	5
Cloud-Native Applications	6
Containerized Applications	7
Serverless	8
DevOps	9, 10
Docker	11, 12
Application Security	13
Network Services in Application Deployment	14
Top OWASP Threats	15

Caution

The goal of self-assessment is to gauge your mastery of the topics in this chapter. If you do not know the answer to a question or are only partially sure of the answer, you should mark that question as wrong for purposes of self-assessment. Giving yourself credit for an answer that you correctly guess skews your self-assessment results and might provide you with a false sense of security.

1. Which characteristic matches a SaaS deployment model?
   a. Provider deploys your software customizations.
   b. Any update to the software requires a new license.
   c. You can recommend tweaks to the underlying infrastructure.
   d. None of the above
2. Which is a good deployment model for real-time IoT sensors?
   a. SaaS model
   b. Edge computing model
   c. Private cloud model
   d. Hybrid cloud model
3. Which statement about application deployment methods is true?
   a. Containers are always the correct choice regardless of the workload.

b. The deployment method should match the workload, since bare-metal, virtualized, cloud-native, containerized, and serverless each fit different needs.

c. Serverless is the right method for long-running, always-on processes.

d. Bare-metal is required for microservice architectures.

4. Which workload is the best candidate for a bare-metal deployment?

a. A function that runs for a few seconds once an hour

b. A big data application that benefits from dedicated, direct access to server hardware

c. A microservice that must scale out automatically under load

d. A web app that must run identically on a laptop and in production

5. Besides improving server utilization, what key benefit did virtualization introduce?

a. It eliminated the need for an operating system.

b. It guaranteed sub-second start times.

c. It made applications portable and no longer tied to a single physical server.

d. It removed the need to protect secrets.

6. Which statement best describes a cloud-native (microservice) application?

a. It runs as a single large process on dedicated hardware.

b. It is distributed across many small services, so the loss of one server doesn't take the application down.

c. It can run only inside a public cloud provider.

d. It stores all application state on the local disk of one server.

7. In which of the following ways are containers different from virtual machines? (Choose two.)

a. Containers have less storage requirements than VMs.

b. VMs can run any operating system, but containers run only on Linux.

c. Containers usually start faster than VMs.

d. VMs are better if you have a microservice architecture.

8. Which deployment method is best for processes that are run periodically?

a. Serverless

b. Containers

c. Virtual machines

d. All of the above

9. What is the second way of DevOps?

a. Automation

b. Continuous learning and experimentation

c. Culture

d. Feedback loop

10. What is continuous integration?

a. Automated software delivery and deployment

b. An Agile software development technique

c. The process of merging development work with the code base for automated testing

d. None of the above

11. A Docker image uses what type of file system?

a. Layered file system

b. NFS

c. XFS

d. Distributed file system

12. What command do you use to launch an nginx container on port 80 of the host file system?

a. **docker image build -p 80 nginx**

b. **docker start -it -d nginx -p 80|80**

c. **docker container run -p 80:80 -d nginx**

d. None of the above

13. Which of the following should NOT be stored in source code or container images?

a. Application logic

b. README files

c. API keys, database passwords, and other secrets

d. UI text strings

14. A reverse proxy in front of a web application most commonly handles which of the following? (Choose two.)

a. TLS termination

b. Database queries

c. Request routing and caching

d. Source code compilation

15. Which of the OWASP Top 10:2025 categories includes both SQL injection and cross-site scripting (XSS)?

a. A01:2025 Broken Access Control

b. A05:2025 Injection

c. A07:2025 Authentication Failures

d. A10:2025 Mishandling of Exceptional Conditions

16. NIST SP 800-145 defines cloud computing through which three lenses?

a. Essential characteristics, service models, and deployment models

b. Public, private, and hybrid clouds

c. IaaS, PaaS, and SaaS

d. Cost, performance, and security

Foundation Topics

Application Deployment Models

The concept of cloud has wormed its way into almost every facet of modern life. It has become central to how we interact with our friends, buy things, watch TV and movies, and run our businesses. So what is it really? Well, that is a very subjective question that has to take into account your own personal perspective. If you asked five people what cloud is, you would probably get eight different answers. Nearly half of all business meetings about cloud, when it was in its infancy, were spent on determining what it was and how it could be used. Is it a service? Is it a product? Is it something magical that will solve all business problems?

The biggest issue with this line of thinking is trying to call cloud a “thing” in the first place. The key value of cloud is that it gives you a new way of operating IT—allowing it to be fast, agile, and able to align better to the goals of the business. From this perspective, cloud can help transform manual and inefficient processes into something the business can use as a competitive advantage and a way to better serve customers. The strength of cloud is realized in the speed and agility it affords as an operational model to get applications deployed and providing value more quickly. Cloud has fundamentally changed how we think about building and operationalizing applications.

Today Amazon, Google, Microsoft, Cisco, and others offer cloud services. Many startups are building their applications and services completely in the cloud. Uber, for example, has completely transformed the transportation industry through a cloud-based application that allows us to easily find someone willing to pick us up and take us where we want to go. Square has transformed credit card processing by allowing anyone with a smartphone and an inexpensive hardware add-on to take credit card payments. The growth of cloud services like these will continue at a steady pace, forcing businesses to rethink how they engage with their customers and operate business-critical services.

NIST Definition

To define cloud, we need a common taxonomy, and for that we can turn to the National Institute of Standards and Technology, a U.S. government agency responsible for standardizing weights, measurements, and technology standards and practices. NIST Special Publication (SP) 800-145 was written to solve the "what is cloud" dilemma. While SP 800-145 is a rather short document, it hits on the major aspects of cloud and is therefore useful as a framework for understanding application deployment options and prepping for the 200-901 CCNAAUTO exam. In SP 800-145, NIST defines cloud through three primary lenses: essential characteristics, service models, and deployment models (see Figure 13-1).

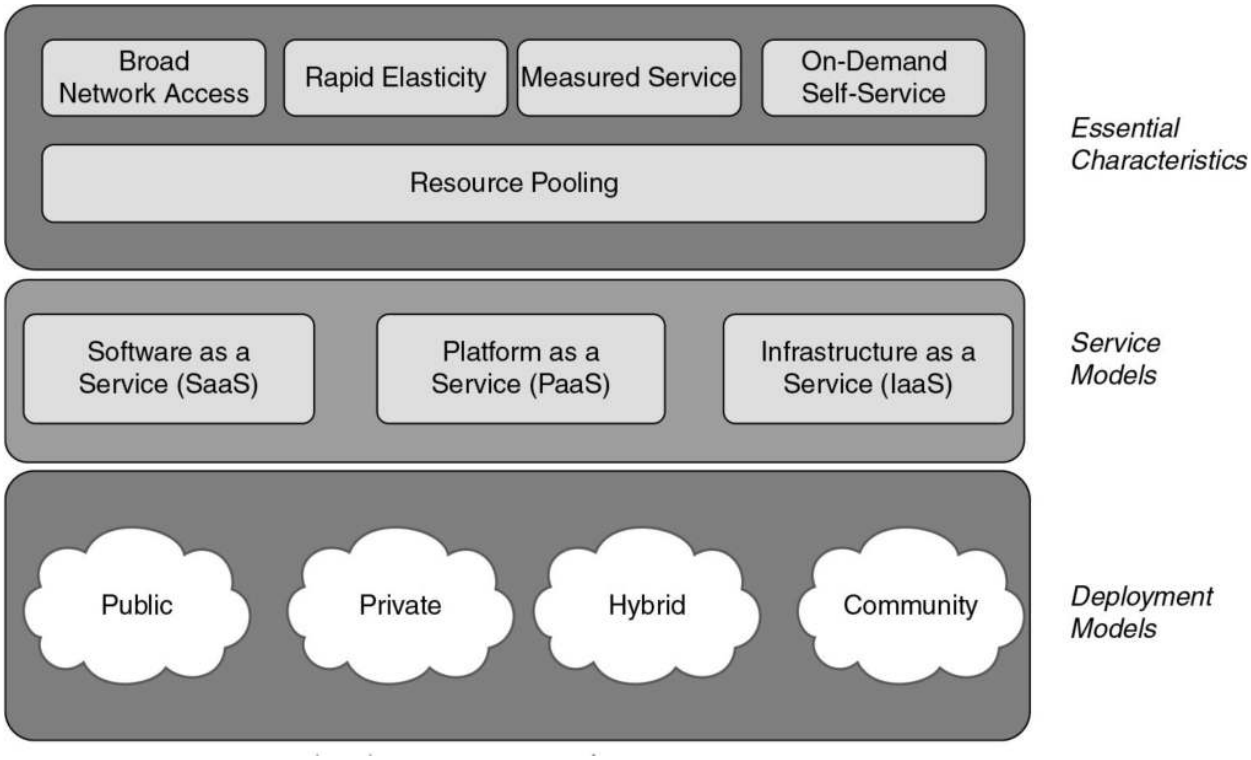


Figure 13-1 *NIST Cloud Computing Definition*

Essential Characteristics

According to SP 800-145, the essential characteristics describe the core requirements of any cloud and are the differentiators that determine whether a service can be classified as being a cloud offering. SP 800-145 defines the essential characteristics as follows:

- **Broad network access:** Services are available over the network and accessed via standard protocols and communications technologies on any type of client device (mobile phone, tablet, desktop, and so on).
- **Rapid elasticity:** Capacity can be automatically provisioned and decommissioned to scale the service to demand.
- **Measured service:** Cloud systems measure resource utilization (compute, storage, and network) and charge for those services accordingly. Utilization is monitored, controlled, and reported on, allowing transparency for the service provider and customer.
- **On-demand self-service:** The cloud consumer can provision compute, storage, and network as needed, without human interaction, through automation or self-service portals.
- **Resource pooling:** The infrastructure is a common pool of resources that can serve multiple customers at the same time. The customer does not interact with the underlying hardware, and workloads can be moved within the cloud environment based on demand without the end user knowing or needing to be involved.

Service Models

The cloud service models mainly differ in terms of how much control/administration the cloud customer has to perform. Figure 13-2 shows the following service models:

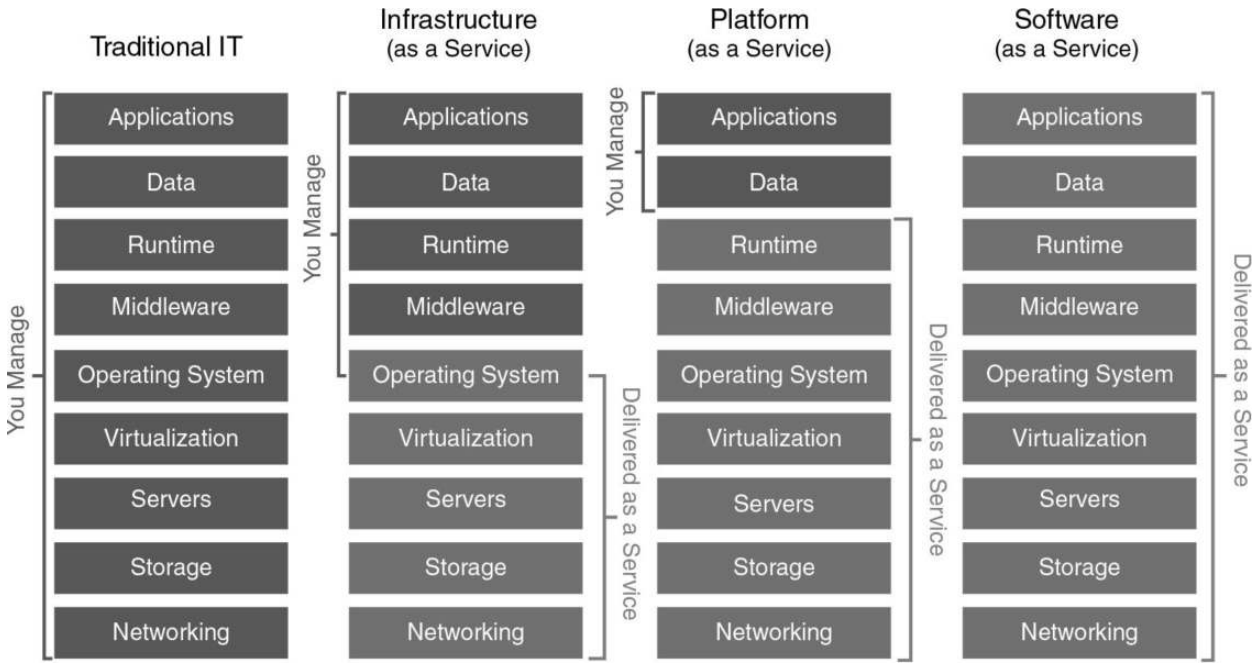


Figure 13-2 *Cloud Service Models*

- **Software as a service (SaaS):** A service provider hosts, manages, and controls an application and offers it to the customer to use. The customer does not interact at all with the underlying infrastructure, operating systems, storage, or network. There may be some customization capabilities, but they are usually limited to application-specific configuration. What you see is what you get.
- **Platform as a service (PaaS):** A PaaS provider supplies a fully integrated service/software suite, and customers can deploy their applications on top of this suite with a predefined set of application programing interfaces, libraries, software development kits, and/or other tools and services. Customers can program the application in any way they choose and can customize it directly to their own workflow, as long as the customization is within the parameters of the service provider's offering. Customers do not have to worry about integration of the underlying infrastructure components, and in the case of a managed PaaS service, have no management responsibilities for the underlying infrastructure.
- **Infrastructure as a service (IaaS):** A provider enables the customer to provision compute, storage, and networking to run any combination of software and operating systems. While customers have no control of the underlying infrastructure hardware platform, they have full control over the software and services they deploy within the cloud service. The customer is also responsible for the maintenance of the software they deploy, including patching and upgrading.

Application Deployment Options

Cloud can be deployed in a number of ways. The deployment model chosen depends on whether you want to own your own infrastructure, rent your infrastructure, or have a mixture of both. There are a multitude of reasons you might choose one option over the others. Factors such as protecting sensitive data, economics, and speed all need to be weighed in making choices between deployment models.

Private Cloud

A private cloud is provisioned for a single organization that may have multiple service consumers within its business units or groups (see Figure 13-3). The organization may own the private cloud or lease it from another entity. It also does not have to reside on the organization's premises and may be in another facility owned and operated by a third party. The following are the key characteristics of a private cloud:

- A private cloud has dedicated resources for a single organization.
- Connectivity can occur through the Internet, fiber, or a private network.
- A private cloud may be on premises or off premises.
- Applications are deployed within a private cloud, and the organization has complete control and responsibility for their maintenance and upkeep.

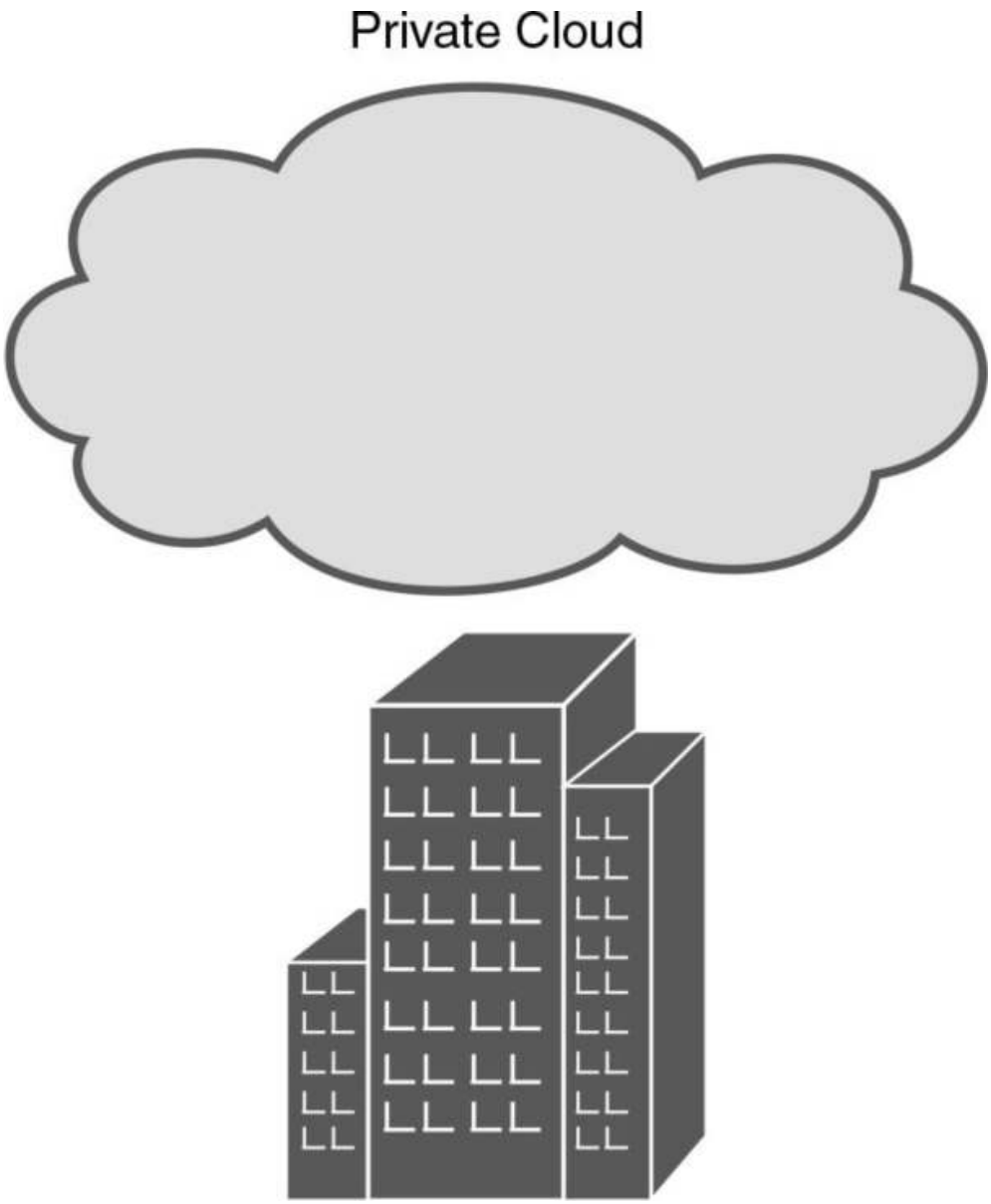


Figure 13-3 *Private Cloud*

Public Cloud

A public cloud is provisioned for open utilization by the public at large (see Figure 13-4). Anyone with a credit card can gain access to a public cloud offering. A public cloud exists solely on the premises of the cloud provider. The key characteristics of a public cloud are as follows:

- Resources are publicly shared.
- A public cloud supports multiple customers.
- Connectivity occurs through the Internet or virtual private connections.
- Use of a public cloud is billed based on usage.
- Applications are deployed on the infrastructure hardware and services offered by the cloud provider. The user is responsible for security and application operations but not responsible for hardware maintenance.

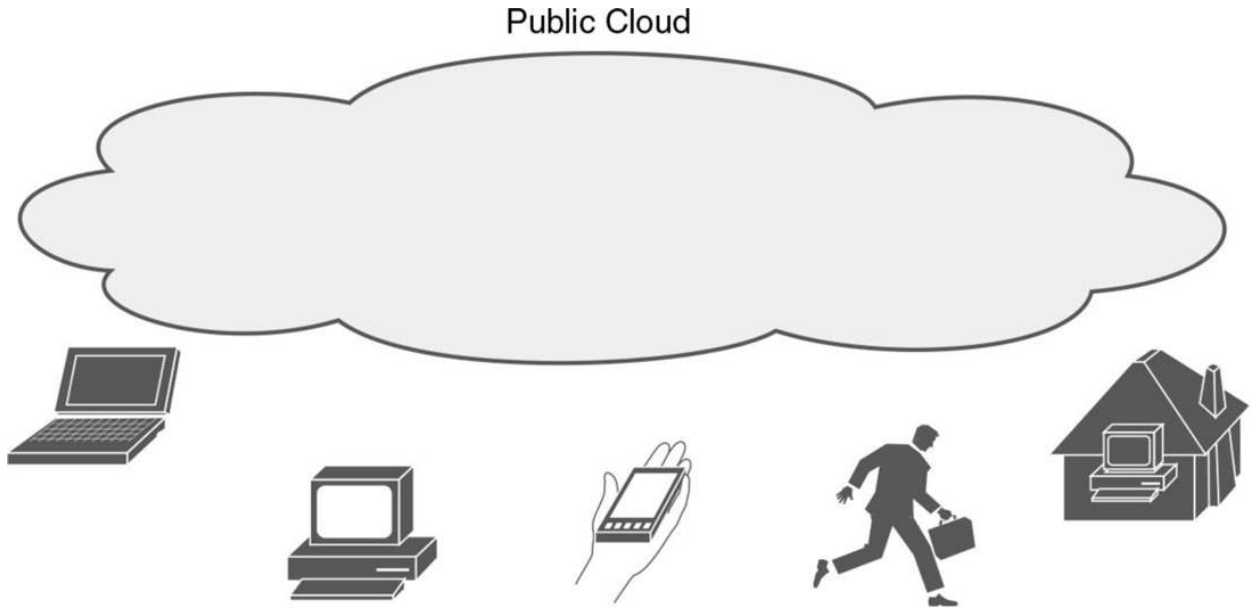


Figure 13-4 *Public Cloud*

Hybrid Cloud

A hybrid cloud is composed of two or more distinct cloud infrastructures (private and public, for example) and is used to extend capabilities or reach and/or to augment capacity during peak demand periods (see Figure 13-5). The key characteristics of a hybrid cloud are as follows:

- A hybrid cloud is a combination of public and private models.
- A hybrid cloud has on-premises and off-premise resources.
- Orchestration between the parts of a hybrid cloud is the responsibility of the application owner.
- Applications are deployed in a distributed fashion, with some services and components residing on premises and others (such as client access) in the cloud provider's environment.

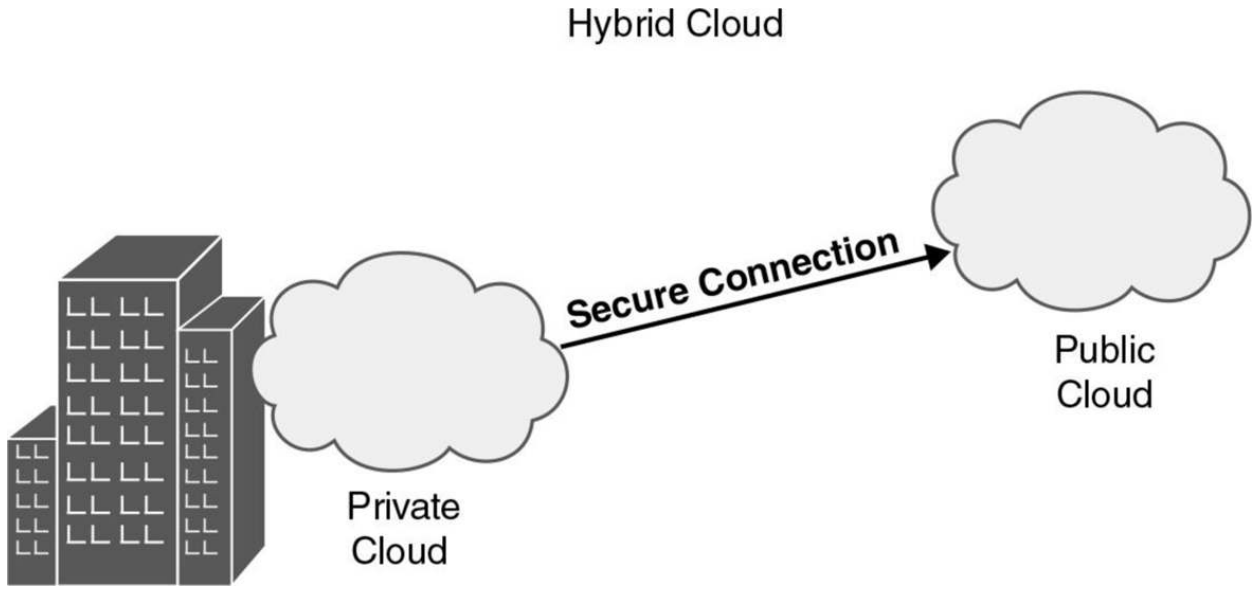


Figure 13-5 *Hybrid Cloud*

Community Cloud

A community cloud, as shown in Figure 13-6, is unique in that it is provisioned for the sole utilization of a specific community of customers, such as a school district or multiple government agencies. Basically any group of entities that have a common policy, security, compliance, or mission can join together and implement a community cloud. The key characteristics are as follows:

- A community cloud is a collaborative cloud effort that shares infrastructure between several similar organizations with common needs.
- A community cloud can be owned, managed, and operated by one or more members of the community or a third party.

- A community cloud may reside on premises or off premises.
- Applications are deployed in a shared fashion and operated by the community for the use of all community members. This can be similar to a hybrid model but with a focus on resource sharing.

Edge and Fog Computing

With the increase in IoT applications, devices, and sensors, the various deployment models covered so far are just not up to the task of handling the sheer volume of data being created and needing to be processed. Self-driving cars, robotics, computer video processing, and many other use cases can produce gigabytes of data in real time. This sea of data cannot be consumed and processed by a centralized cloud application, given the cost of transporting the data and the latency involved in receiving information back to do anything with it. Your self-driving car would run into a ditch before a cloud service could detect and respond.

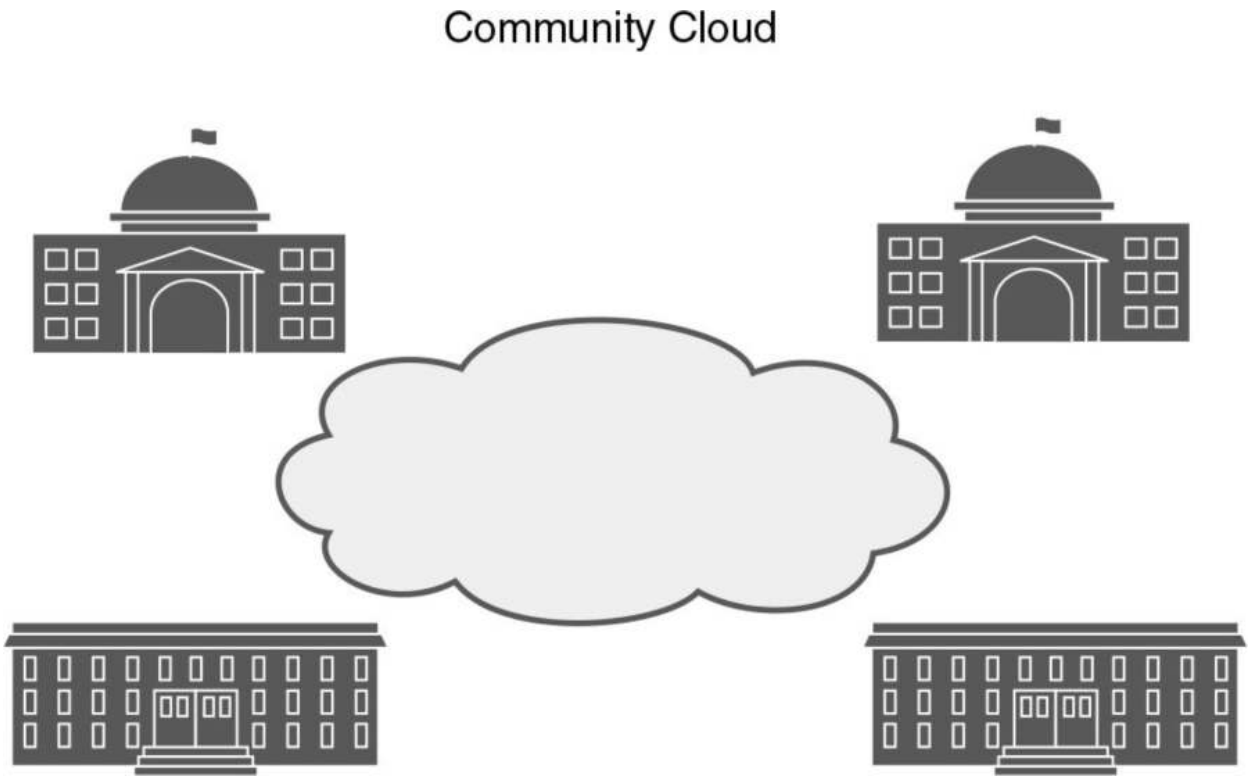


Figure 13-6 *Community Cloud*

These constraints required engineers to think differently about use cases requiring intelligence at the edge. A new class of application deployment was created, called fog computing (because fog is basically a cloud on the ground). The idea behind fog is to leverage the scale and capabilities of cloud models to handle the control and deployment of applications at the edge. With edge computing, you let local processing of data occur closer to the actual sensors and devices where it can be acted on more quickly. Think of fog as a framework and structure that enables edge computing. Figure 13-7 shows how edge and fog work together.

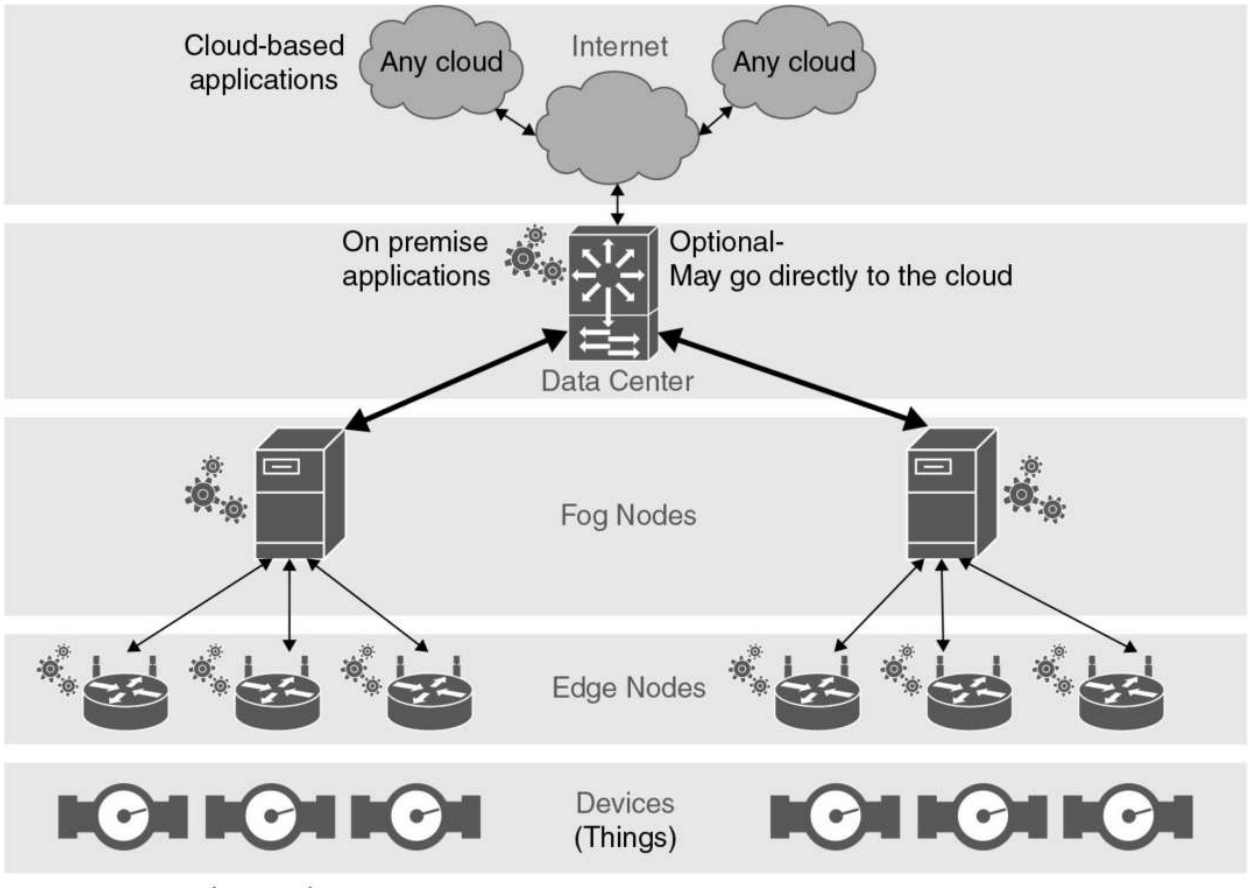


Figure 13-7 *Edge and Fog Computing*

Application Deployment Methods

What is IT's value to the business if you boil it down to the simplest aspect? IT is charged with supporting and maintaining applications that the business relies on. No one builds a network first and then looks for applications to stick on it. Instead, a network is built to connect applications to employees and customers. The importance of the application and its evolution have driven many of the advances we have seen in cloud.

Bare-Metal Application Deployment

The traditional application stack, or bare-metal, deployment is fairly well known. It's how the vast majority of applications have been deployed over the past 40 years. As Figure 13-8 shows, one server is devoted to the task of running a single application. This one-to-one relationship means the application has access to all of the resources the server has available to it. If you need more memory or CPU, however, you have to physically add new memory or a new processor, or you can transfer the application to a different server.

While bare-metal application deployment is an effective way to isolate applications and get consistent performance characteristics, it's very inefficient. In fact, many of these traditional server deployment models result in enormous amounts of waste in regard to power and cooling of servers when they are not under load. It's not uncommon to find a server during non-peak hours using less than 10% of its capacity, which is not very cost-effective.

Some applications, however, really need a dedicated server. Big data applications benefit greatly from a dedicated server environment; most Hadoop clusters consist of numerous one-rack-unit servers all running in parallel. These workloads consume 100% of the resources of the server, making them poor candidates for virtualization.

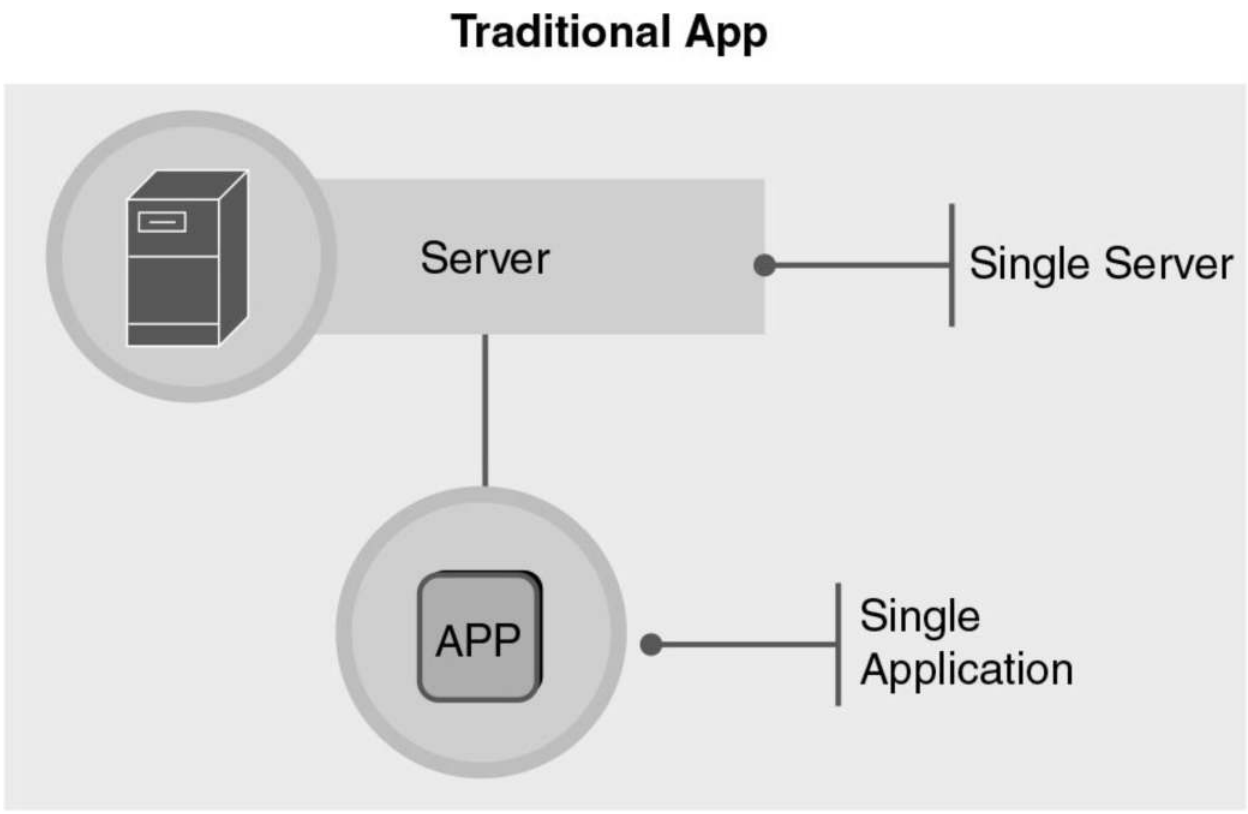


Figure 13-8 *Bare-Metal Application Stack*

Virtualized Applications

Virtualization was created to address the problems of traditional bare-metal server deployments where the server capacity was poorly utilized. The idea with virtualization is to build one large server and run more than one application on it. Sounds simple, right? Many of the techniques mentioned earlier, from the days of the mainframe, were leveraged to create an environment where the underlying server hardware could be virtualized. The hypervisor was created to handle all of the time slicing and hardware simulation. This made it possible to run various applications and operating systems at the same time and reap the benefit of better utilization of resources. Figure 13-9 shows this concept.

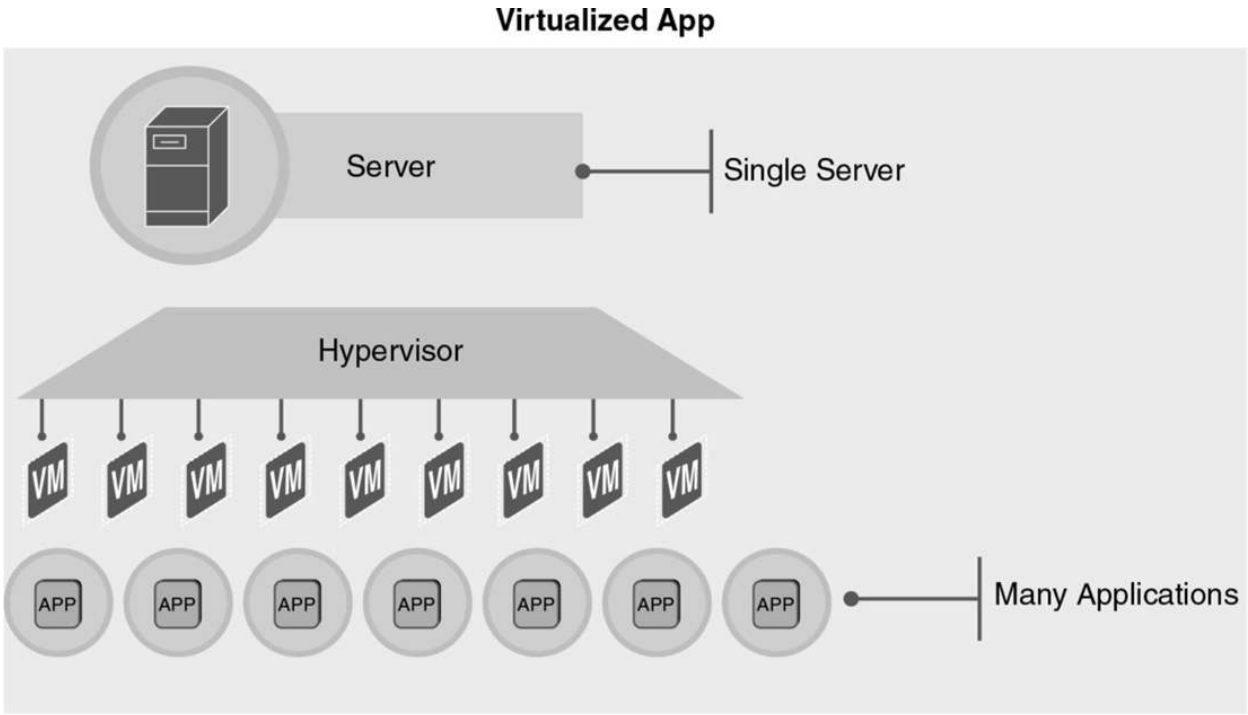


Figure 13-9 *Virtualized Application Stack*

Virtualizing hardware had other benefits, too. The ability to make applications portable and not tied to a single server allows for mobility in the data center. If a server needs to be worked on, a compatible live-migration platform can move the running virtual machine to another server with little or no application downtime. You can also create common deployment packages for new virtual machines and applications; this gives administrators the ability to quickly provision a new server in a consistent and secure manner.

The virtualized application is a key aspect of cloud, but *virtualization* and *cloud* are not synonymous. A cloud environment could easily be configured to deploy a physical server as well as a virtual machine. Conversely, just because you have a virtualized infrastructure doesn't mean you have a cloud; the big difference is in the operational model.

Cloud-Native Applications

Cloud-native applications, which may use microservices, represent the further intersection and evolution of virtualization and a cloud environment. As virtualization became popular, application developers started to ask themselves why they needed a heavy operating system to run their applications. All the patching, drivers, and configuration requirements don't go away just because you load a virtual machine. All that work still needs to be done. What if applications could be decoupled and written as simpler packages that do not each need a full guest operating system in order to function? That's exactly what happened. As Figure 13-10 shows, cloud-native applications can run across multiple servers.

If a server goes bad in a cloud-native application environment, a resilient application can continue operating because the application is distributed across many servers in the data center, and the application was written from the beginning to handle failures and redirect work to working nodes. If you need more capacity, add a new server, and automation can add it to the server pool. The infrastructure deploys the application onto the new server, and capacity increases after workloads are scheduled and started. If load decreases, the orchestration platform can scale down unused nodes automatically to save money. Large-scale cloud applications such as Facebook and Twitter operate in this fashion, which is why they appear to have infinite capacity from a user perspective.

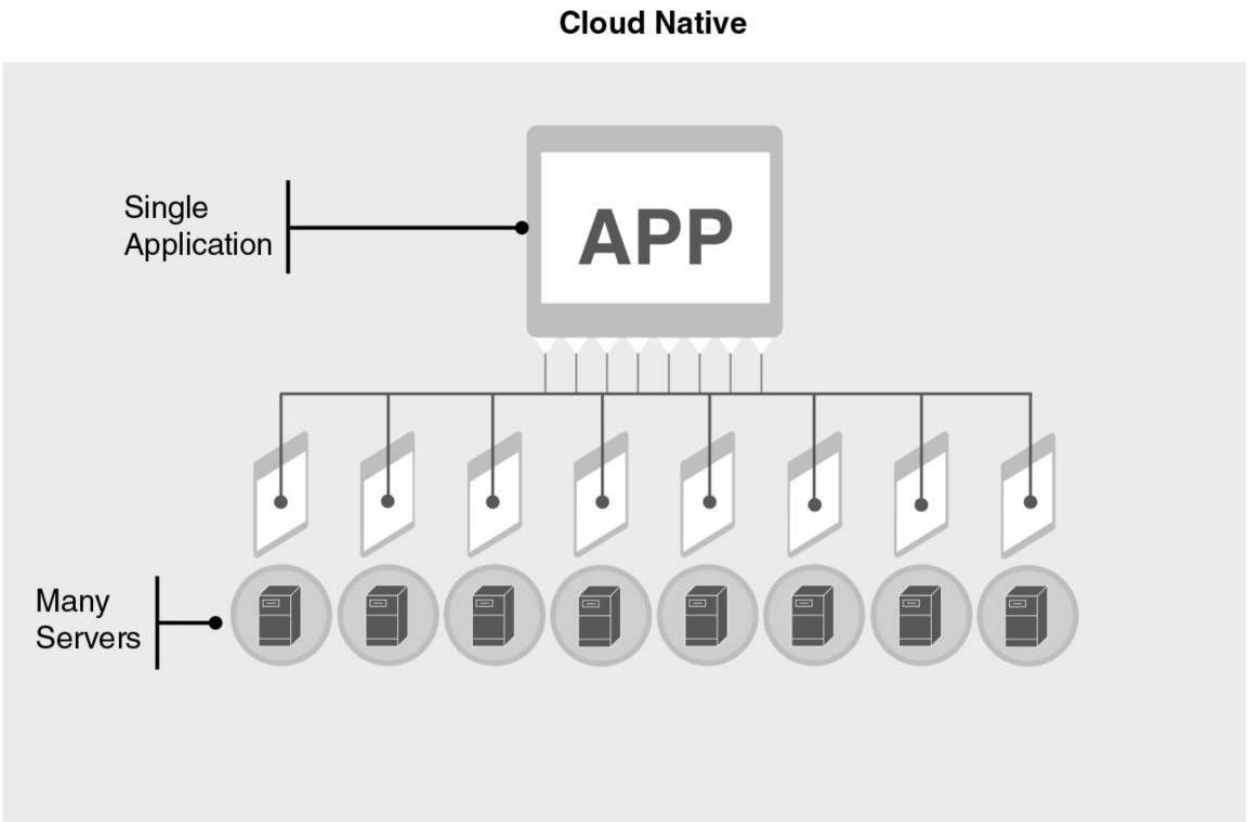


Figure 13-10 *Cloud-Native Applications*

Containerized Applications

The evolution of applications to leverage microservices architectures was a very important change that provided developers with new ways to build massively scaled applications with tremendous improvements in terms of availability and fault tolerance. Microservices made available small custom-built components, but they didn't address the need for an efficient infrastructure that could support them and their dynamic nature. Containers became a very popular way to provide an easier way to get application components deployed in a consistent way. Imagine being able to have all of the components and dependencies of your application loaded in an interchangeable format that you can simply hand off to operations to deploy. This was what containers offer. Google

created Kubernetes and made it open source to provide an orchestration and automation framework that can be used to operationalize this new application deployment model.

The benefits of containers are as follows:

- Consistency for deployment automation
- Simplified lightweight image files measured in megabytes (whereas VM files are measured in gigabytes)
- Providing only what the app needs and nothing else
- Ability to work the same in production as on a developer's laptop
- Open community-built best-of-breed containers for faster innovation
- Ability to deploy applications in seconds

Deploying applications in a container decouples parts of the application. Consider a traditional application stack like LAMP (Linux, Apache, MySQL, Perl/Python/PHP). This was the foundation for many websites over the years. To deploy a LAMP stack in a virtual environment, you load an operating system and application on top of a hypervisor and then repeat this process for however many virtual machines you need to support the number of users you expect. Include a load balancer in front of these virtual machines, and you can add capacity as needed. The problem with this design is that the virtual machines run full operating systems, replicated a number of times. You also have to maintain the virtual machines the same as you would as if they were physical servers. Capacity is handled manually, and you can easily run out of resources if your website receives more visitors than you planned for. Figure 13-11 shows a LAMP deployment on virtual machines.

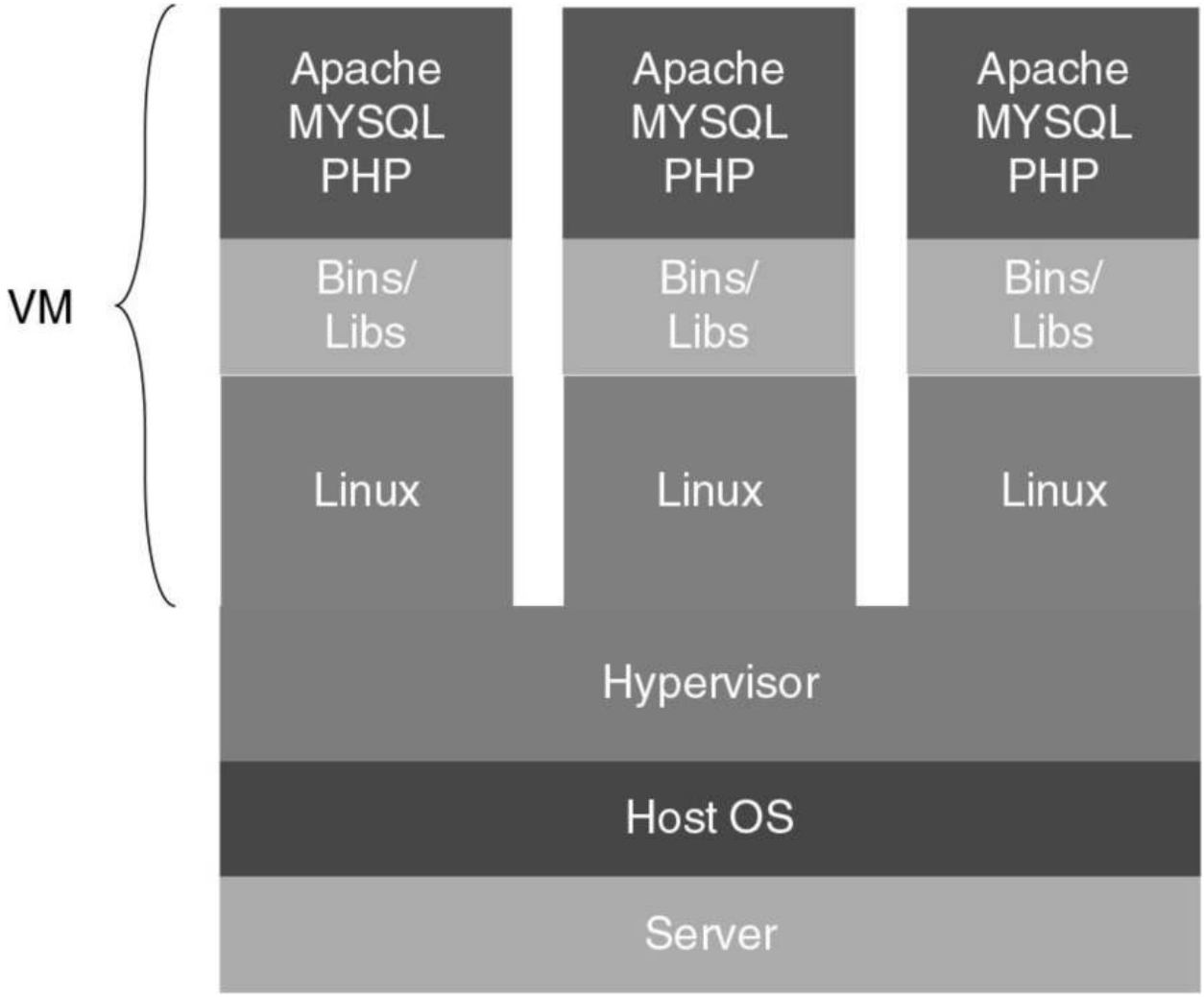


Figure 13-11 *LAMP Stack on a Virtual Machine Infrastructure*

Now consider what happens if you deploy that same application stack in a container infrastructure. You decouple the components of the stack and deploy them separately. So instead of everything being loaded on a single virtual machine, you break off the Apache web server and PHP code from the MySQL database. By themselves, they are much smaller in size and resource utilization and can be scaled independently of each other. Instead of having a gigabyte or more being taken up for an operating system and hardware virtualization of memory and other parts of virtualization, the container simply shares the kernel of the underlying operating system of the server it is running on. The container is treated like an application and is run in an isolated

space within the operating system. Any binaries and libraries needed to run the container are prepackaged, which means you don't have to touch the container host server. Figure 13-12 shows a containerized LAMP stack.

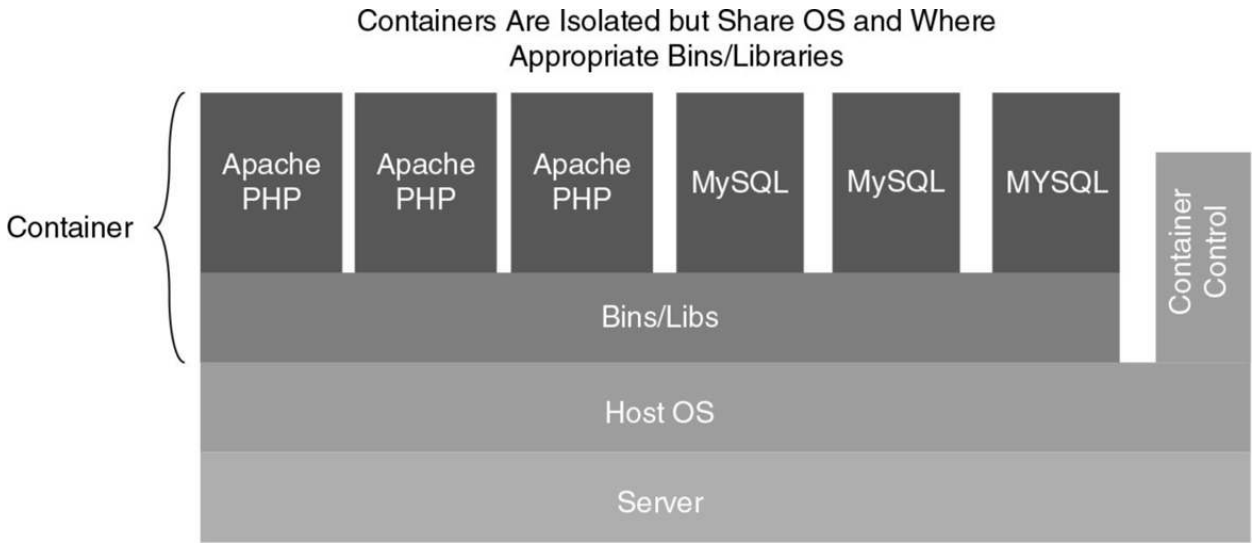


Figure 13-12 *Containerized LAMP Stack*

Docker excels at running containers on a single host, but production workloads often need orchestration across many hosts. Kubernetes is a widely used platform for that. It schedules containers across a cluster, restarts failed ones, scales up and down, and handles networking and storage. Docker Swarm and HashiCorp Nomad are alternatives. A lot of teams skip running their own orchestrator entirely by using a managed service like Amazon ECS, Google Cloud Run, or Azure Container Apps.

Serverless

Serverless is one of the silliest names in the industry. Of course, there is a server involved! However, this type of application deployment mechanism is intended to make it even easier to get an application up and running, and the term *serverless* conveys this idea. You simply package and deploy your code to a serverless service, and the infrastructure runs your code without managing the underlying servers. A common serverless model is function as a service (FaaS), which supports event-driven, request-driven, scheduled, and batch workloads. When writing code, you create a function that you call repeatedly at different places in the code to make it more efficient. Serverless deployment uses the same write once/use many times concept: You write some code and then invoke it in response to an event or request.

Serverless applications typically execute some type of application logic but are usually stateless and use external services to store data. The function or calling application accesses those services as needed. For example, say you want to create a function that converts images you receive to a universal size and shape. You can then load this function into a serverless offering from AWS, and any time you receive a new picture, it is sent to the function and returned in the correct format (see Figure 13-13).

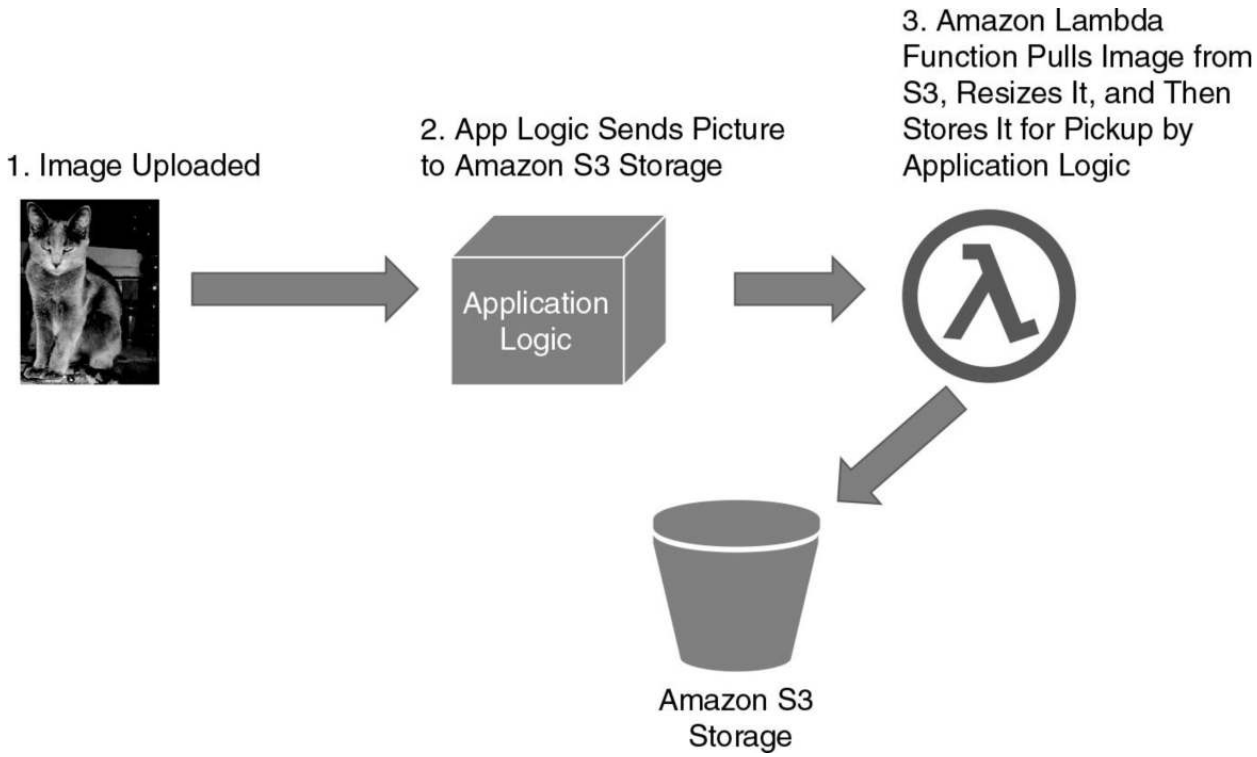


Figure 13-13 *Serverless Image Resizing Example*

The whole idea behind serverless and why a customer may want to use it is that it doesn't require dedicated hardware or resources to be set aside for these periodic processes. The customer pays for the resources used, and then the services are shut down until needed again; this is a better cost model. The following are some of the advantages of serverless deployment:

- **Cost:** Serverless can be significantly cheaper than prepaying for infrastructure that is underutilized. The pay-as-you-go computing model means on-demand execution is not billed while the function is idle, although provisioned capacity can incur charges.
- **Scalability:** Developers don't need to build in their own autoscaling policies or technologies. The provider is responsible for adding more capacity on demand.
- **Easier to use and write code for:** Small teams of developers can use serverless deployment without needing to involve infrastructure teams or acquire advanced skills.

The following are some of the disadvantages of serverless deployment:

- **Latency:** Spin-up time from idle for the function can cause significant delays that must be accounted for in application design.
- **Resource constraints:** Some workloads need more resources than the service may typically be allocated for, causing heavy charges, which may make it cheaper to dedicate infrastructure.
- **Monitoring and debugging:** There is limited visibility into the underlying infrastructure, which makes it difficult to troubleshoot performance issues with the application. You are often limited to the tools provided by the service provider, which are likely to be proprietary.
- **Security and privacy:** Most providers are built on proprietary offerings and use shared resources. Misconfiguration can result in compromises and data loss, just as with any other cloud service. There are on-premises options for serverless that can give businesses more control over their data and security posture.
- **Vendor lock-in:** This is a big one. Each provider has its own tools and frameworks, which makes it very difficult to switch providers if costs go up or if services are not working as planned. Migration between providers is not trivial.

DevOps

Agile has dramatically changed the software development landscape by introducing a more efficient and faster way of delivering software and value to the business. Much of the improvement in the development process has been focused on delivery speed. You can have as much Agile development as you want, but if you can't get the software deployed in a timely manner, you can't take advantage of its capabilities. Software was traditionally built and tested by developers, and if it worked on their laptops or test systems, it was pitched over the fence to IT operations for deployment. When things did not go as smoothly as planned, a significant amount of finger-pointing would ensue. Developers would claim that the software worked fine in testing on their laptops, so the infrastructure must be broken. The operations teams would say that the buggy and poorly written software was

stressing their environment and creating all kinds of extra work. These two groups were in need of some serious marriage counseling.

Developers and operations teams often have very different expectations and metrics for success. Developers care about functional code, APIs, libraries, services, and Agile sprints. Success to them means software works on their laptops and in testing, and they finish their latest sprints in time. Operations, on the other hand, cares about the environment being stable, standards, templates, and not getting awakened at 2 a.m. to go fix a problem. Success to operations means software and applications are stable, backup and restore processes work, and all systems are operating within defined thresholds. Developers are all about creating new features and capabilities in their software; change is a constant in their world. Operations folks are looking to maintain stability in their environment. The two could not be further apart.

If you take a calendar view of the two organizations, it becomes very apparent where the challenge is (see Figure 13-14). How can you possibly deploy new features and capabilities added every 2 weeks as part of an Agile sprint when your operational maintenance window is only open every 6 months?

Traditional IT service delivery is slow, manual, and often prone to errors (see Figure 13-15). The infrastructure is a shared resource, and one change can have a ripple effect that could break other unrelated systems. The fragile nature of the infrastructure makes it very hard to anticipate issues that can arise. To combat this, many IT organizations create layers upon layers of approval processes. While this sounds like a rational way to protect the business from downtime, the net effect is to slow new deployments to a crawl. For these reasons and many others, DevOps was created.

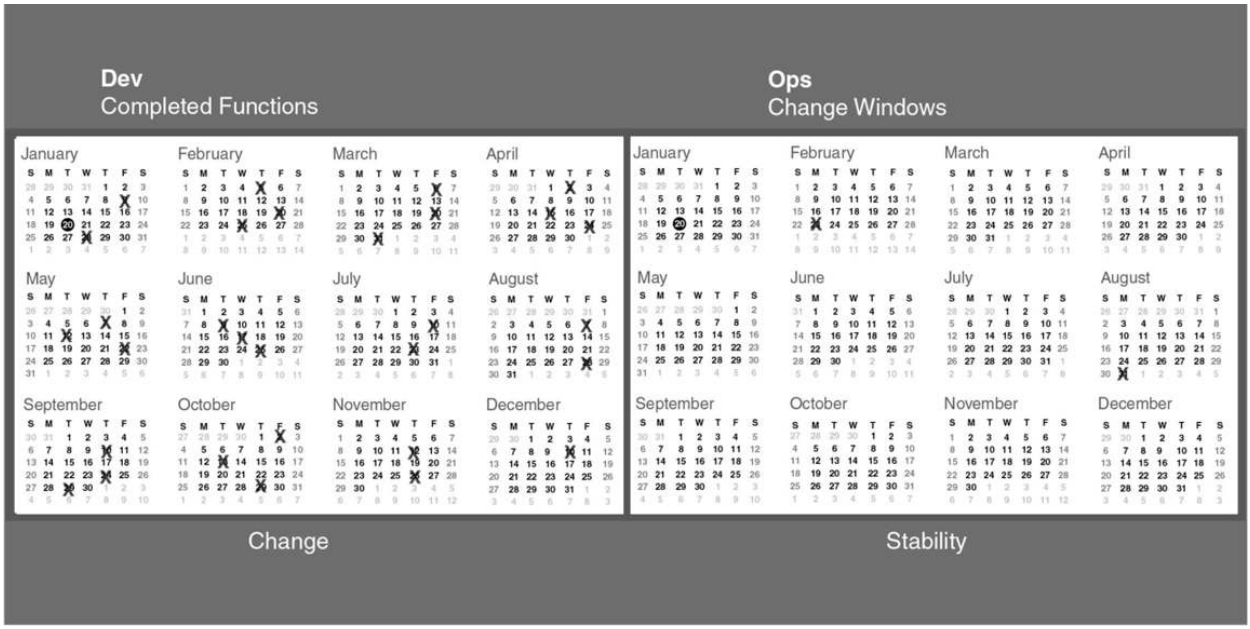


Figure 13-14 *Dev and Ops Calendars*

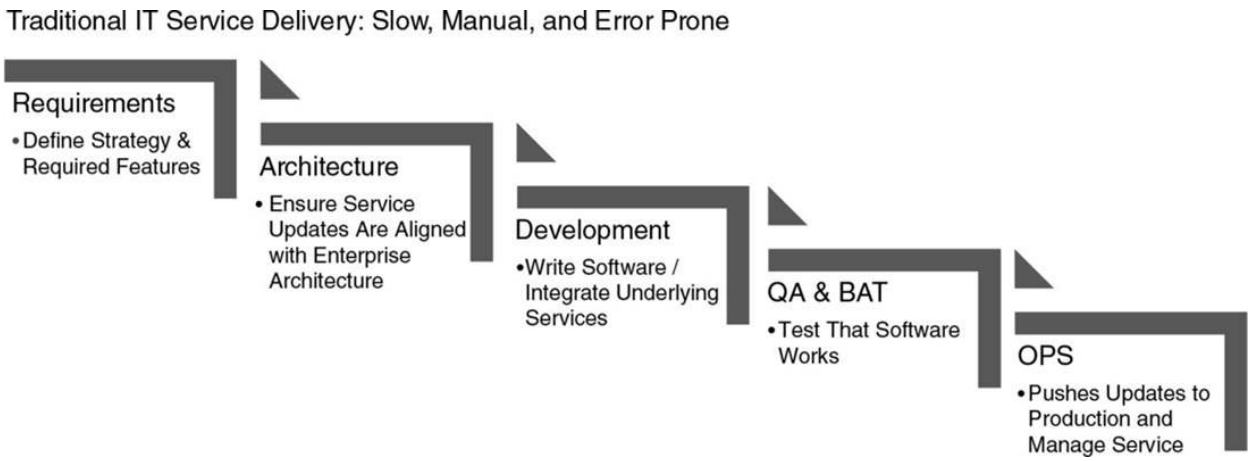


Figure 13-15 *Traditional Sequential Approach to Operations*

What Is DevOps?

In 2009 two employees from Flickr (an image sharing site), John Allspaw and Paul Hammond, presented a talk titled "10+ Deploys per Day: Dev and Ops Cooperation at Flickr" to a bunch of developers at the O'Reilly Velocity conference. In this talk, Allspaw and Hammond said that the only way to build, test, and deploy software is for development and operations to be integrated together. The sheer audacity of being able to deploy new software so quickly was the carrot that fueled the launch of

the DevOps concept. Over the years since then, DevOps has moved from being revered by a small group of zealots and counterculture types to being a very real and quantifiable way to operate the machinery of software creation and release. The vast majority of companies that do software development are looking for ways to capture the efficiencies of this model to gain a competitive edge and be better able to adapt to change from customer and industry perspectives.

In a nutshell, DevOps is a culture of sharing in which developers and operations folks are one unit that can rise and fall together. It is a practical application of Lean and Agile. The goal of DevOps is to be a real-time business enabler by removing wasted effort and bureaucracy from getting in the way of better addressing the needs of the customer. DevOps has five guiding principles:

- **Culture:** For DevOps to work, organizational culture must change. This is by far one of the most difficult aspects to embrace, but it is the single most important factor for success. DevOps requires a culture of sharing.
- **Automation:** While DevOps is more than just a set of software tools, automation is the most easily identifiable benefit. Automation techniques greatly speed up the deployment process, enable defects to be caught and corrected earlier, and eliminate the need for human intervention in repetitive tasks.
- **Lean:** Reducing wasted efforts and streamlining the process are the goals of Lean. It's a management philosophy of continuous improvement and learning.
- **Measurement:** Unless you measure your results, you can never improve. Success with DevOps requires the measurement of performance, process, and people metrics as often as is feasible.
- **Sharing:** DevOps requires a culture of feedback and sharing. Breaking down silos and creating an inclusive shared fate environment is the ultimate goal.

Figure 13-16 shows the core components of DevOps and how they are interrelated.

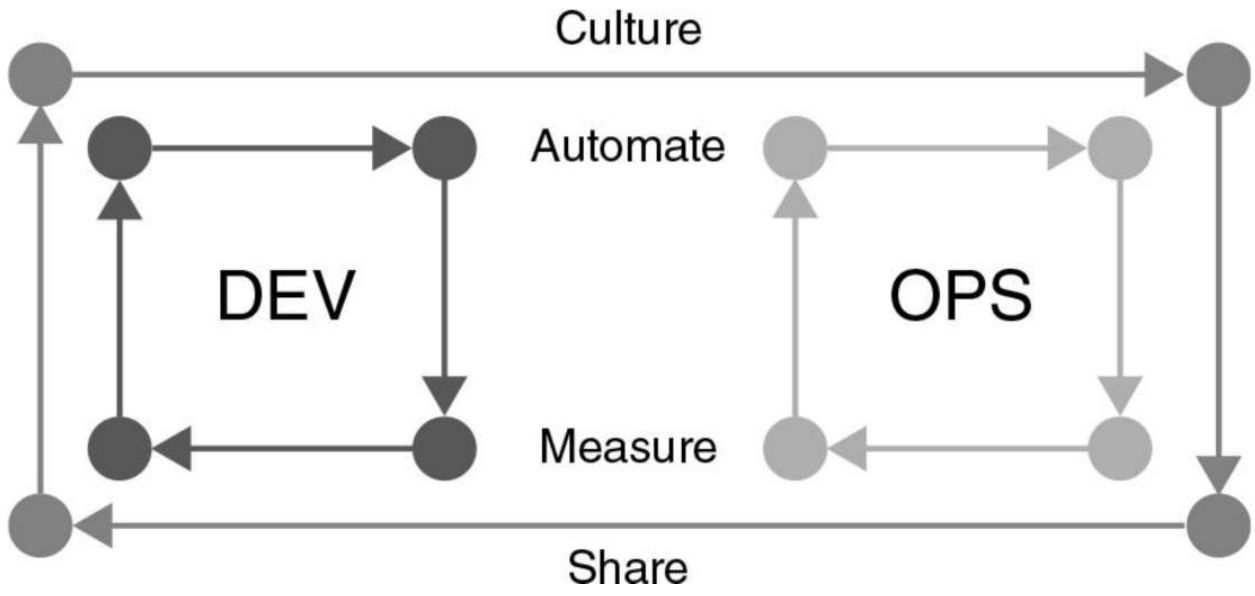


Figure 13-16 *DevOps*

Putting DevOps into Practice: The Three Ways

When discussing DevOps, it's hard not to mention two very important books on the subject. The first of these books, *The Phoenix Project*, by Gene Kim, Kevin Behr, and George Spafford, set the stage for understanding the benefits of DevOps practices through a story that builds on the experiences of the authors and contributors and shows how a fictional company moves from being almost out of business to a rising star outpacing its competition. The interesting thing about the book is that you can find many examples of your everyday life and practices outlined in it. While it probably won't be made into a blockbuster movie anytime soon, it is a quick read that easily shows the benefits of DevOps practices. The second important DevOps book is *DevOps Handbook: How to Create World-Class Agility, Reliability, and Security in Technology Organizations*, written by Gene Kim, Jez Humble, Patrick Debois, and John Willis. This book is a more practical view of the subject and provides insight into the "Three Ways" of DevOps (discussed in the following sections), offering a framework for implementation in your own organization. If you want to know more about DevOps, you should certainly pick up these two books.

First Way: Systems and Flow

A highway system at full capacity is basically a parking lot, where no one is moving. We have all had to spend time staring at brake lights in traffic. For a technologist, an equivalent work scenario featuring overtime and missed family dinners occurs all too

commonly. Many problems occur because of the ripple effect of not knowing how everything flows and not being able to visualize all work moving through the system. Having a bird's-eye view of how work flows from development to production is essential to figuring out optimal ways to get to the end goal of a working application that provides value. Many teams use kanban boards to visualize work, whether through an application like Jira or by simply putting sticky notes with the workflow on a whiteboard so that everyone can see what is being done and what needs to be done. Using a kanban board can help a team be more rational in what it expects to accomplish in a given period of time. Figure 13-17 shows the first way of DevOps, which is focused on systems and flow.

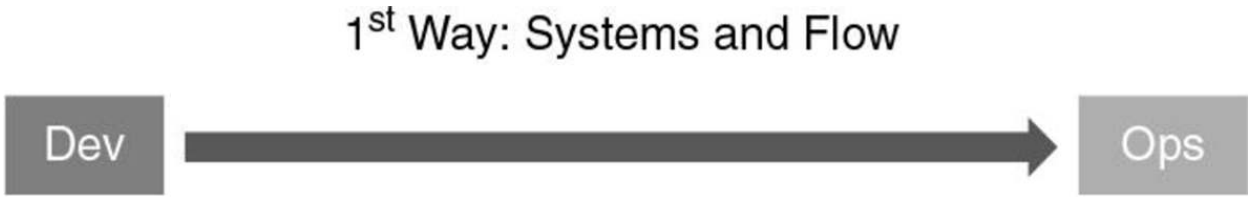


Figure 13-17 *First Way: Systems and Flow*

To accomplish work goals, you have to reduce work units, or batches of work, so that they are more streamlined to prevent workload traffic jams. In an Agile sprint, it is common to work on a small set of features during a two-week period to reduce how much is being worked in a given interval of time. This allows developers time to build the capability without being overwhelmed or pulled in multiple directions. This also applies to the operations side, as both sides are integral to success. Developers can't just rain app updates on their operations teams without having a solid view of how each update gets deployed and the operations teams have to build a system that is able to handle the speed and agility developers need. By having both developers and operations work together to understand where constraints are, you can optimize the system and move faster.

Quality is not a bolt-on at the end of the development and deployment process. It has to be integrated into the whole system. Waiting for a problem to be found in QA will create compounded issues that can make things much worse. The earlier you address a challenging situation, the more your overall work effort will be reduced. This equates to saving money and time. The whole point is to reduce wasted time on rework.

You have to continually optimize the system to make it more efficient. Nothing is ever perfect with DevOps; you are in a constant state of removing inefficiencies. You must be able to adapt to business needs and your customers' ever-changing requirements.

The following are the key characteristics of the first way:

- Make work visible
- Reduce batch sizes
- Reduce intervals of work
- Build in quality by preventing defects from being passed downstream
- Constantly optimize for business goals

Second Way: Feedback Loop

One of the things you will see repeatedly in DevOps is analogies to manufacturing. Since so much of this management philosophy is derived from Lean and the Toyota Production System, concepts like defect prevention are front and center in the approach. The idea of a feedback loop is to provide guidance and direction on what is working and what is not working. If something isn't working, you have to make sure that it doesn't happen again so you don't end up on a hamster wheel of pain. DevOps requires that you take feedback as the gift that it is and make corrections. Figure 13-18 shows the second way of DevOps, which is focused on the feedback loop.

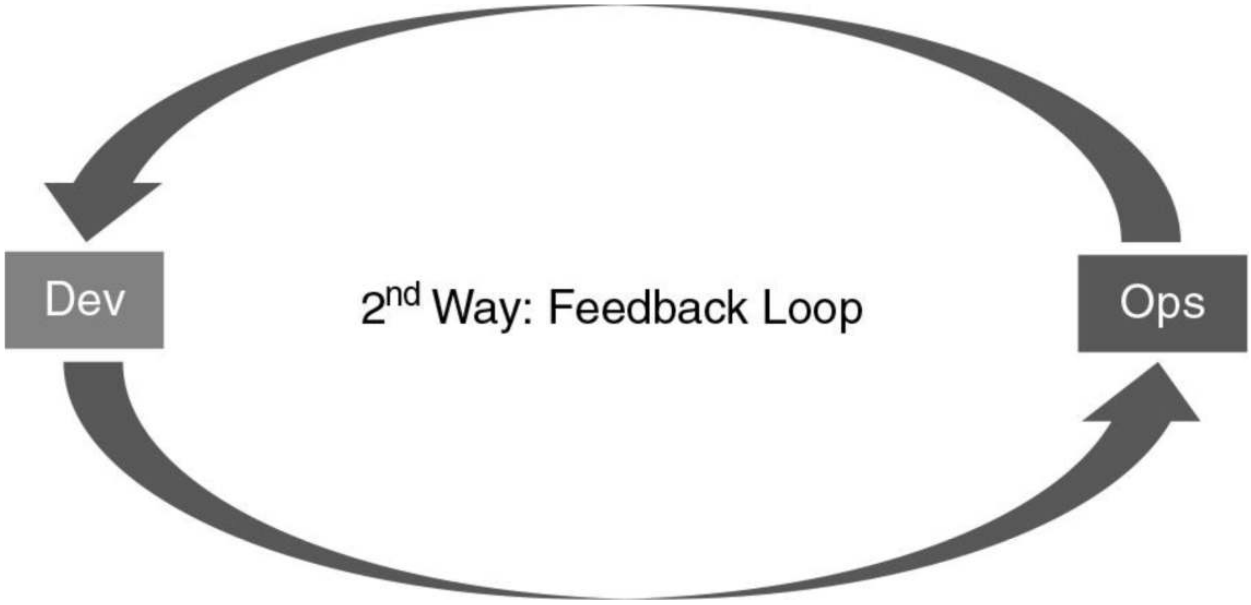


Figure 13-18 *Second Way: Feedback Loop*

The more you sample feedback, the faster you can detect issues and recover from them. In the Toyota Production System, the manufacturing line has something called an Andon Cord, which can be pulled by anyone at any time to halt the production line. This form of feedback is immediate and highly visible. It also kicks off the process of swarming a problem until it is fixed, where everyone in the area focuses on solving the issue before the line is restarted. In DevOps, this type of empowerment means that when problems with software are detected, the whole team pitches in to fix the underlying cause.

Once a problem has been overcome, it is not just time to have a party; it is time to document and improve the processes that led up to the issue at hand: Learn from your mistakes so you can move on.

The following are the key characteristics of the second way:

- Amplify feedback to prevent problems from happening again
- Enable faster detection and recovery
- See problems as they occur and swarm them until they are fixed
- Maximize opportunities to learn and improve

Third Way: Continuous Experimentation and Learning

How do you foster innovation? Create a culture of trust where your people are allowed to try new things and fail. That doesn't mean that you are just rolling the dice on crackpot ideas; rather, it means having a dynamic and disciplined approach to experimentation and risk taking. It's very rare that success happens on the first try. When problems occur, avoid pointing fingers and blaming people; instead, figure out what went wrong and what could be improved to make it better. Figure 13-19 shows the third way of DevOps, which focuses on continuous experimentation and learning.

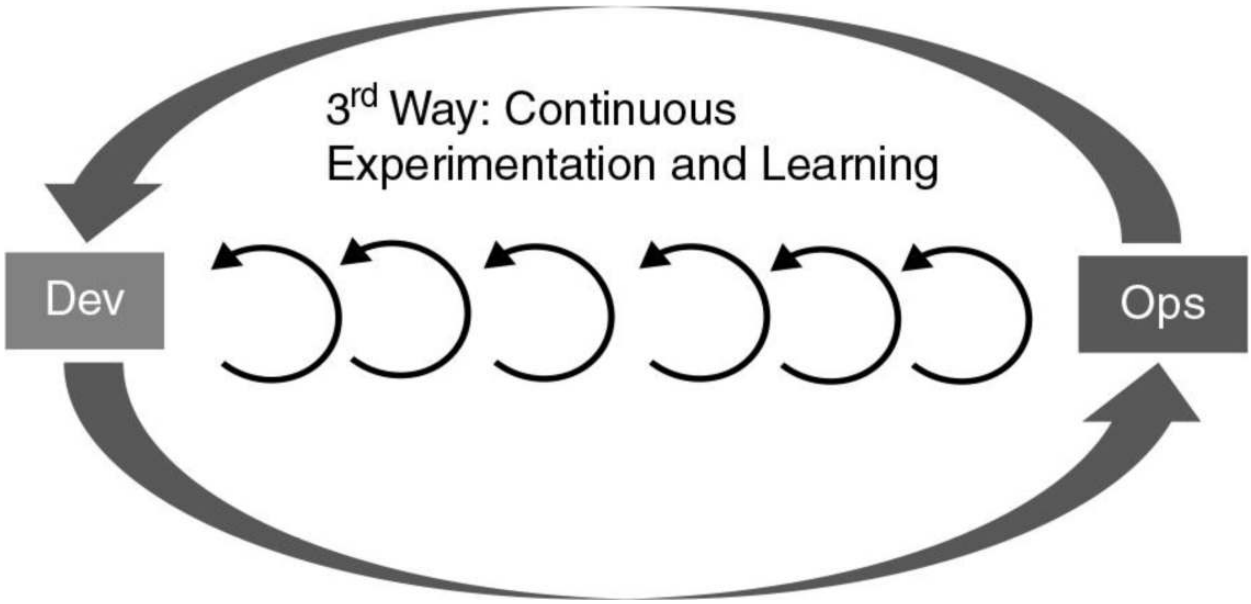


Figure 13-19 *Third Way: Continuous Experimentation and Learning*

DevOps talks a lot about continuous improvement and fixing issues to make the system perform better. It's a noble goal, but when do you find time? The simple answer is that you need to make time. In other words, you schedule time to get people focused on improving the system and try new methods and technologies.

Finally, you must build a culture of sharing and learning. A simple method to accomplish this is to create shared code repositories and get those treasure troves of information off individual laptops and available to the rest of the organization.

The following are the key characteristics of the third way:

- Conduct dynamic, disciplined experimentation and risk taking
- Define time to fix issues and make the system better
- When things go wrong, don't point fingers
- Create shared code repositories

DevOps Implementation

The "feedback loop" of DevOps depends on knowing what's actually happening in production. Modern practice has crystallized around the three pillars of observability: logs (what happened), metrics (how much, how fast), and traces (how requests flowed through services). OpenTelemetry (OTel) is the open-source observability framework and set of specifications for instrumenting applications to emit all three; backends like Grafana, Datadog, New Relic, and Honeycomb consume that data and turn it into dashboards and alerts.

> **Note**
>
> AI is increasingly woven into deployment, monitoring, and incident response. AIOps platforms use machine learning on observability data to detect anomalies, group related alerts, and suggest root causes faster than humans can. AI-assisted code review flags risky pull requests; LLM-driven runbooks help on-call engineers triage incidents at 3 a.m.; infrastructure-as-code generators produce starter Terraform, Kubernetes manifests, and Dockerfiles from plain-English prompts. None of this is tested on the CCNAAUTO exam, but it's part of the day-to-day reality of modern operations teams in 2026. As discussed in Chapter 3, treat AI output as a starting point you read, verify, and shape. Not as the answer.

Implementing DevOps technical practices within an organization typically involves three stages. These stages follow a natural progression that leads to a fully automated code-to-deployment operational model:

- **Continuous integration:** This stage involves merging development work with the code base constantly so that automated testing can catch problems early.
- **Continuous delivery:** This stage involves keeping every successful change ready for production release, usually with a manual release decision.
- **Continuous deployment:** This stage relies on continuous integration and continuous delivery to automatically release code into production as soon as it is ready.

A vast number of technical tools can be used to automate a DevOps environment, from commercial applications to bleeding-edge open-source projects. You need to be aware of this DevOps tool chain if you are asked to administer a DevOps environment. Digital.ai is a company that specializes in DevOps and continuous delivery, and it is a great resource for understanding the various tools in a DevOps pipeline. Its Periodic Table of DevSecOps Tools provides an interactive view of these tools and provides links and descriptions of the tools and technologies you might encounter (see Figure 13-20). You can go to https://digital.ai/learn/devsecops-periodic-table/ to get access and download your own copy.

If you are not interested in the do-it-yourself nature of connecting and configuring open-source tools, you might be better off looking to a vendor that offers a platform as a service or one that will build your DevOps pipeline for you and also offer support with maintenance and upgrades. Regardless of the DevOps platforms or tools you use or whether you build your own or buy them, there are some commonalities for all DevOps pipelines. It's easiest to understand the workings of a DevOps pipeline by seeing it in action. Figure 13-21 is a graphical representation of what happens at each step of a sample DevOps pipeline.

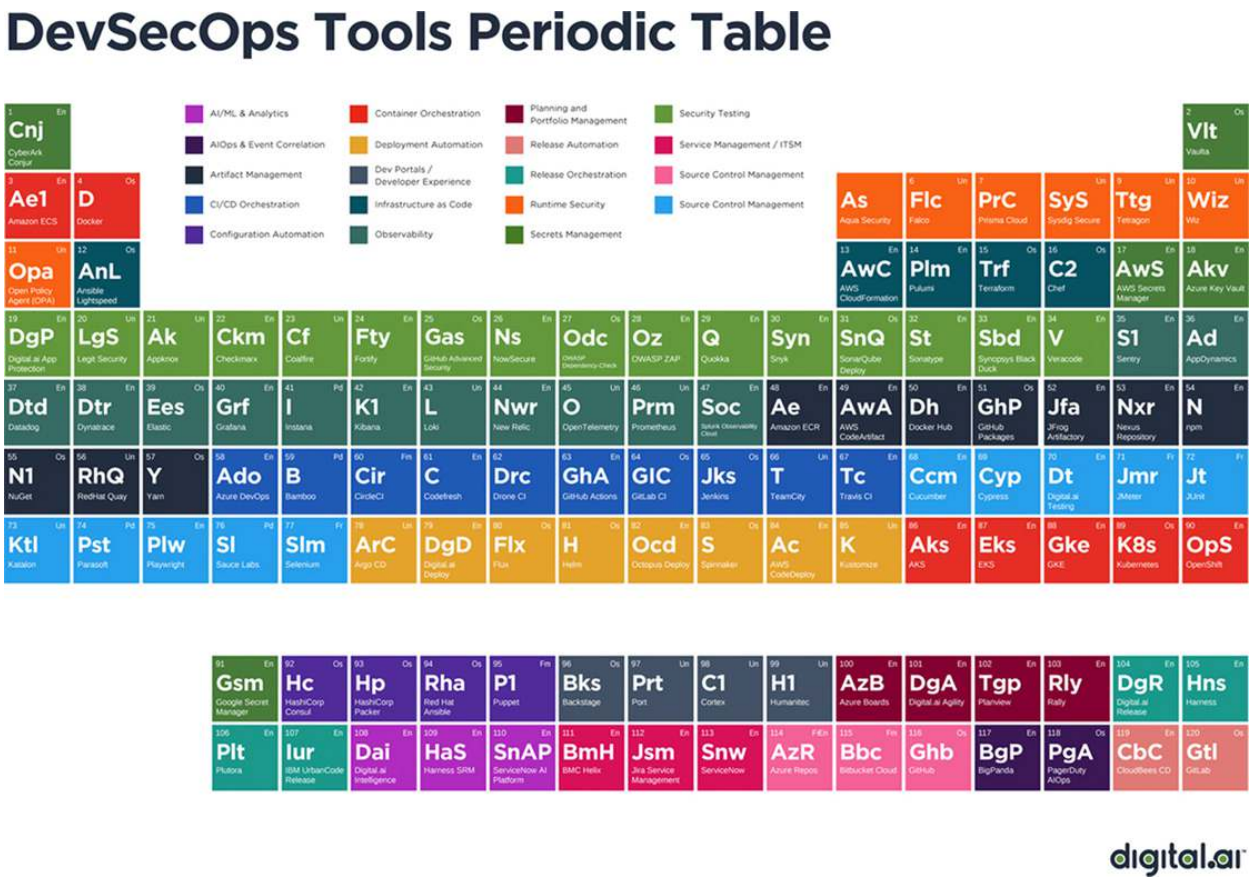


Figure 13-20 *Periodic Table of DevSecOps Tools (Source: https://digital.ai/learn/devsecops-periodic-table/)*

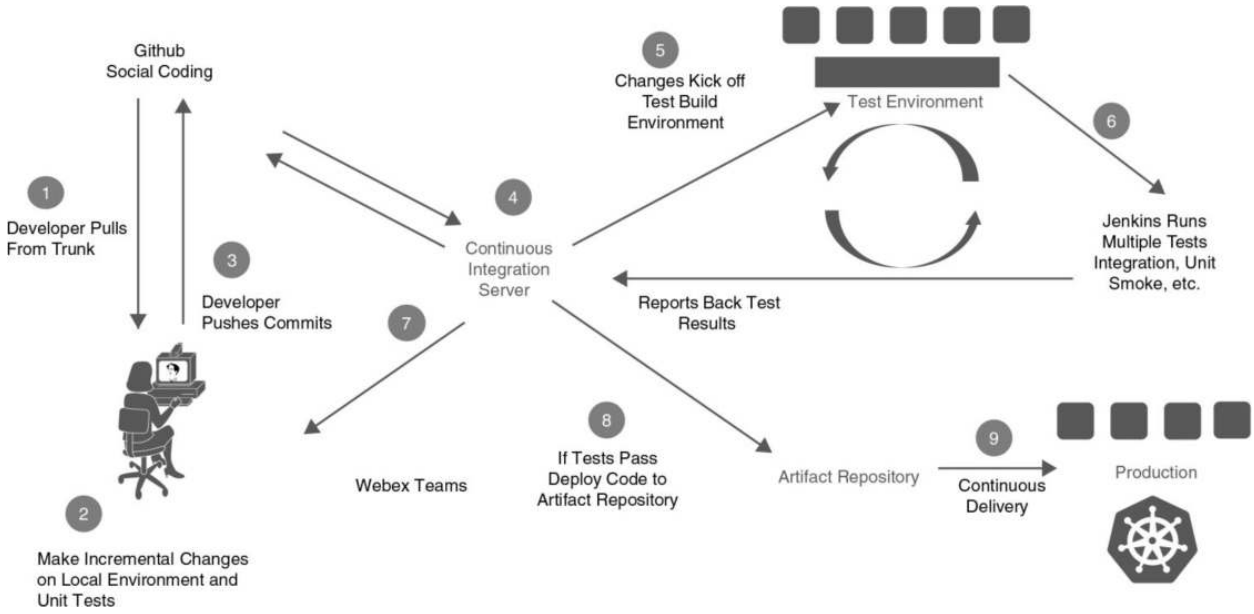


Figure 13-21 *DevOps Pipeline in Action*

Here is how the pipeline works:

Step 1. The developer pulls the latest code from the version control system with Git. This ensures that the developer has the most recent changes and is not working with an old version.

Step 2. The developer makes changes to the code, adding new features or fixing bugs. The developer also writes test cases that will be used to automatically test the new code to software functional requirements. The developer eventually stages the changes in Git for submission.

Step 3. The developer uses Git to push the code and tests to the version control system (for example, GitHub), synchronizing the local version with the remote code repository stored in the version control system.

Step 4. The continuous integration server, such as Jenkins, uses a GitHub webhook or polling trigger to detect new code submissions. When Jenkins is notified of a new commit, it is able to pull the latest version and starts an automated build

process using the test cases.

Step 5. Jenkins kicks off the automated build of a testing environment for the new software build. It is possible to use Python scripts, Ansible, or other infrastructure automation tools to build the testing environment as close to production as possible.

Step 6. Jenkins executes various automated testing, including unit testing, integration testing, smoke testing, stress testing, and security testing. When all the tests are finished, the results are sent back to Jenkins for compilation.

Step 7. Jenkins sends the test results to the developer for review. They can be sent via email, but a more modern way of alerting the developer is through a collaboration tool such as Webex. Jenkins has plug-ins for all major team management tools and allows for easy integration. If the build fails, the developer can use links to the information on what failed and why, make changes to the code, and restart the process.

Step 8. If the build process is successful and all of the tests pass, Jenkins can deploy the new code to an artifact repository (just a fancy name for the place finished software is stored). The software is now able to be deployed in production.

Step 9. Jenkins signals the infrastructure that an updated version of software is ready. For example, there may be a new container ready to be deployed into a Kubernetes platform. The updated container replaces the existing containers with the new code fully tested and ready to go. Now the application is updated with minimal (or no) disruption.

The nuts and bolts of building a DevOps pipeline are beyond the scope of the 200-901 CCNAAUTO exam. There are simply more tools than grains of sand at the beach, and luckily you are not going to be tested on all of them. For technologists, it's easy to focus on the tools and technology, but honestly that's the easy part of DevOps. Changing culture and implementing Lean methodologies are where many companies have struggled. Focus on streamlining your processes and implement technologies that help accelerate your efforts.

CI/CD Pipeline Components

The Steps 1 through 9 above walk through one concrete pipeline, but it's worth pulling out the components that show up in essentially every CI/CD pipeline so you can recognize them in any tool.

The pipeline stages you'll see almost everywhere are:

- Source control: a Git repository (GitHub, GitLab, Bitbucket) that triggers the pipeline on a push or merge.
- Build: compile or package the code, or build a container image.
- Test: run automated tests (unit, integration, sometimes end-to-end). A failing test stops the pipeline.
- Artifact storage: store the build output in a registry like Docker Hub, GitHub Container Registry, Artifactory, or Nexus.
- Deploy: ship the artifact through environments (dev, staging, production), often with approval gates between stages.
- Monitor and feed back: track production health and feed any problems back into the next iteration.

The line between continuous delivery and continuous deployment is just whether the deploy step is automatic. Continuous delivery means every successful build is ready to deploy, but a human still pushes the button. Continuous deployment removes the manual step. Every successful build automatically lands in production.

Pipeline orchestrators are abundant. Common ones include Jenkins, GitHub Actions, GitLab CI/CD, CircleCI, and Argo Workflows. The orchestrator coordinates the stages and pulls in specialized tools (test runners, scanners, registries, deployment controllers) along the way.

> **Note**
>
> GitOps is a deployment pattern that uses a Git repository as the single source of truth for the desired state of your infrastructure and applications. Tools like Argo CD and Flux watch the repository and automatically reconcile the running environment to match. Pull requests become deployment requests; rollbacks become git revert. It's now a common way teams deploy to Kubernetes.

Docker

Some of the best innovations come from reusing older technologies and bringing them forward in a new way. That's exactly what Solomon Hykes and Sebastien Pahl did back in 2010 when they started a small platform as a service (PaaS) company called dotCloud. They were looking for a way to make it easier to create applications and deploy them in an automated fashion into their service. They started an internal project to explore the use of some interesting UNIX technologies that were initially developed in the 1970s to enable process isolation at the kernel level. What would eventually become Docker started as a project to use these capabilities to grow the PaaS business. In 2013, the world was introduced to Docker, which represented a new paradigm in application packaging and deployment that took off exponentially. While dotCloud eventually went away, Docker has grown into the leading container platform. Luckily for us, it was open source from the very beginning, and its containerd runtime component is hosted by the Cloud Native Computing Foundation. While there are other container tools and runtimes, such as Podman, CRI-O, and LXC, several are now widely deployed.

Docker is the most familiar way to work with containers, but it's no longer the only one. containerd and CRI-O are container runtimes that Kubernetes uses directly (Kubernetes removed its built-in dockershim integration with Docker Engine in v1.24 in 2022). Podman offers a Docker-compatible CLI without a daemon and is popular on Red Hat systems. These tools work with standardized container images, defined by the Open Container Initiative (OCI). So a "Docker image" is generally OCI-compatible, and it works with any compliant runtime.

Understanding Docker

Docker containers use two capabilities in the Linux kernel: namespaces, which provide isolation for running processes, and cgroups, which make it possible to place resource limits on what a process can access. These features allow you to run a Linux system within another Linux system but without needing to use virtualization technologies to make it work. From the host operating system's perspective, you are just running another application, but the application thinks it is the only application that is running. Instead of needing to virtualize hardware, you just share the kernel; you don't need to load a full operating system, drivers, and memory management processes each time you want to run an application.

Namespaces

Namespaces are essential for providing isolation for containers. Common namespaces used for this purpose include:

- **mnt (mountpoints):** This namespace is used for mapping access to host operating system storage resources to the container process.
- **pid (processes):** This namespace is used to isolate the process ID number space.
- **net (networks):** This namespace is responsible for network access and mapping communication ports.
- **ipc (interprocess communication):** This namespace isolates System V IPC objects and POSIX message queues, including message queues, semaphores, and shared memory.
- **uts (hostname):** This namespace controls host and domain names, allowing unique values per process.
- **user (UIDs):** This namespace is used to map unique user rights to processes.

Figure 13-22 shows a Linux host using namespaces to isolate three different containers.

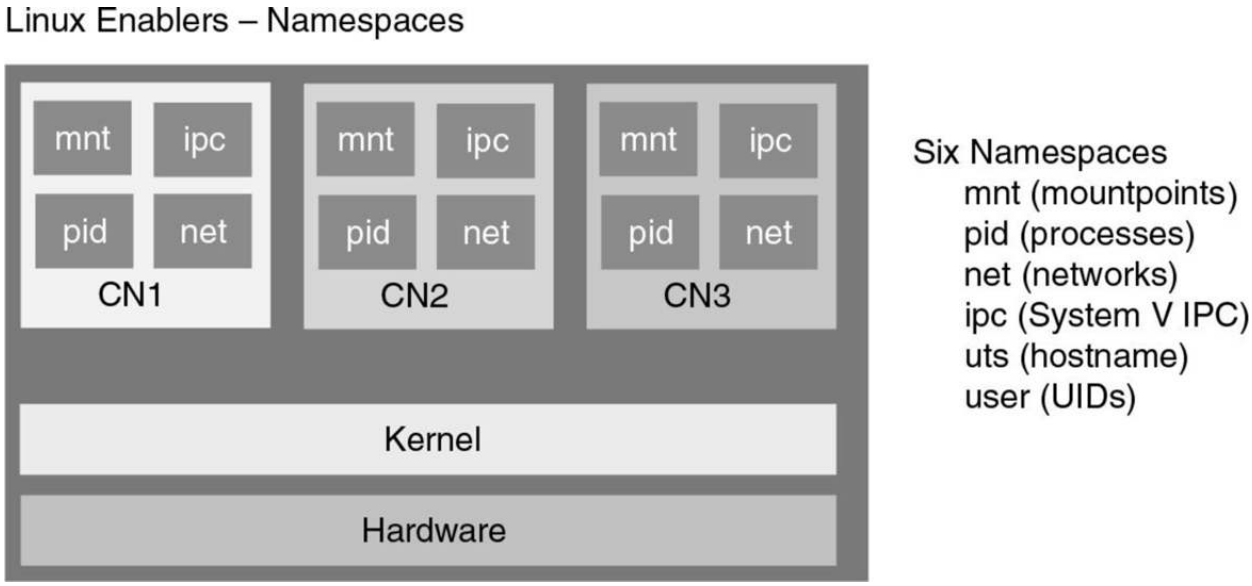


Figure 13-22 *Linux Namespace Isolation for Containers*

Cgroups

Cgroups, or control groups, are used to manage the resource consumption of each container process. You can set how much CPU and RAM are allocated as well as process counts and storage I/O. Each parameter can be managed to tweak what the container sees and uses. These limits are enforced by Linux kernel cgroup controllers. Figure 13-23 shows an example of a cgroup that allocates a maximum of 25% of the various hardware resources of the container host.

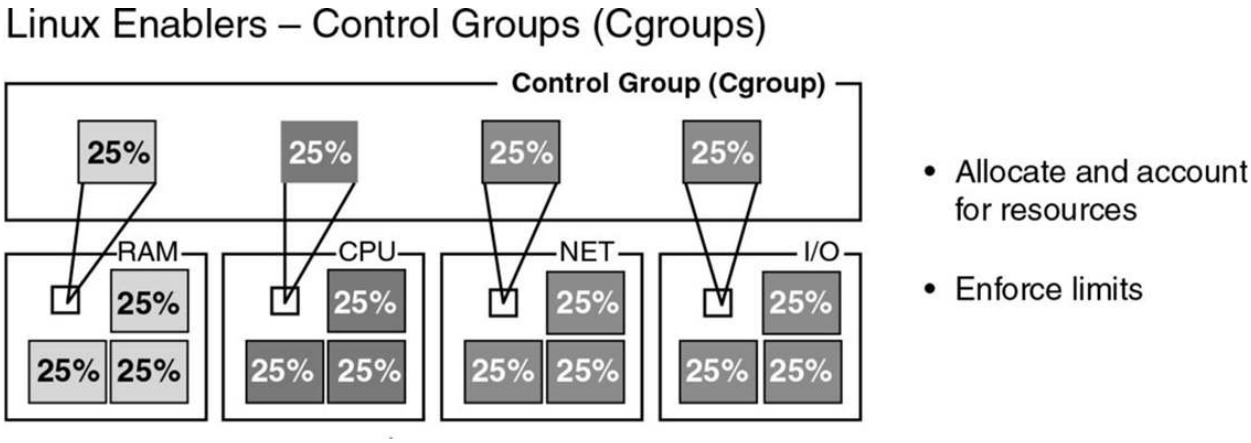


Figure 13-23 *Control Group in Action*

Layered Container Storage

Layered storage is a foundational building block for container images. Docker storage drivers present stacked image layers as one merged filesystem view. If you have ever worked with photo editing software, this will probably make sense to you. Say you want to remove someone or something from a picture (like that ex you never speak of). You take the base picture and then add layers on top of it, and you add or remove pixels until you have a whole new picture. If you want to remove layers you created, you simply peel them back to expose the previous image. Every layer you add on top takes precedence over all the layers below it. In essence, this is how a layered filesystem view works, and it is one of the main reasons such a file system is so efficient at storage. Each layer includes only what is new, and nothing is duplicated from a previous layer. The image layers are read-only, which means they are locked in place. There is a read/write layer that sits on top that you can use to interact with the file system, but changes persist in that container's writable layer until the container is removed. For durable data, use a volume; image updates are made by rebuilding the image. Copy-on-write copies changed files into the container's writable layer. Figure 13-24 shows an example of a container image and its layers.

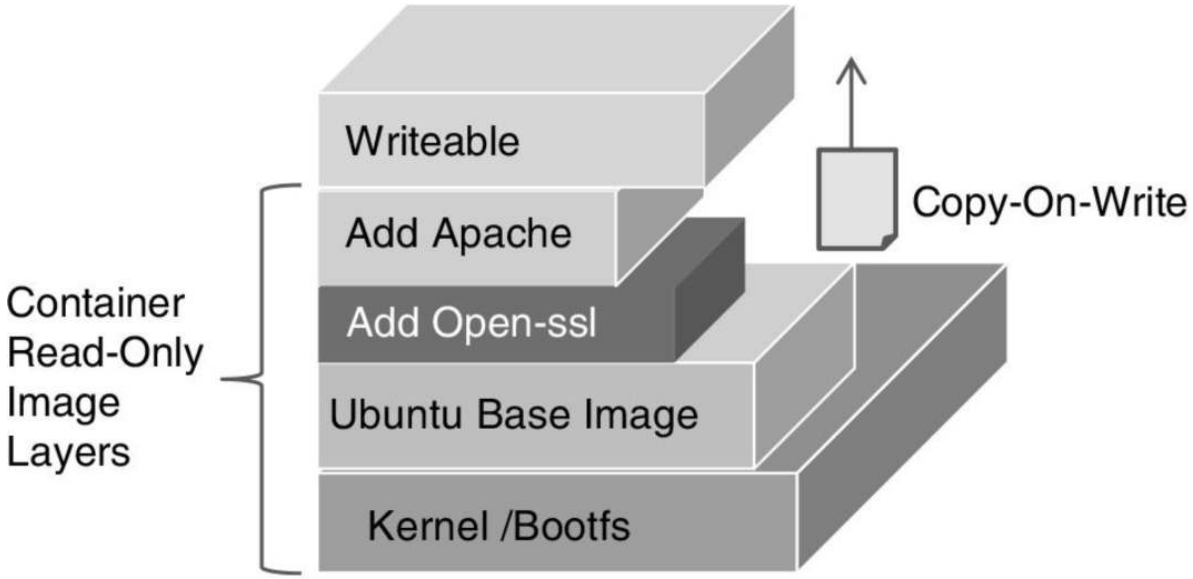


Figure 13-24 *Container Image Layers and a Writable Container Layer*

These read-only image layers are crucial to understand. You may also hear the term *immutable* used to describe the image; it simply means unchanging over time. In other words, deployed containers are normally replaced with containers created from a rebuilt image. In Figure 13-24 you can see an Ubuntu Linux base image layer that has OpenSSL added on top of it via another layer. Above that is the Apache web server. Think of these layers as simply running the **apt-get** command and adding a new software package. What happens if you need to update the OpenSSL layer because of a vulnerability? You rebuild the image with the latest versions of software. Rather than modifying the running container in place, you replace it with a container created from the rebuilt image. Later on in this section, you will learn how this works through a Dockerfile, which is how Docker builds container images.

Docker Architecture

Just like your engine in a car, the Docker Engine is the central component of the Docker architecture. Docker is a client/server application that runs natively on Linux or through Docker Desktop on macOS and Windows and provides all the tools to manage a container environment. It consists of the Docker daemon and the Docker client. Docker allows you to package up an application with all of its dependent parts (binaries, libraries, and code) into a standardized package for software development.

The Docker architecture consists of three primary parts: the client, the Docker host, and the docker registry (see Figure 13-25).

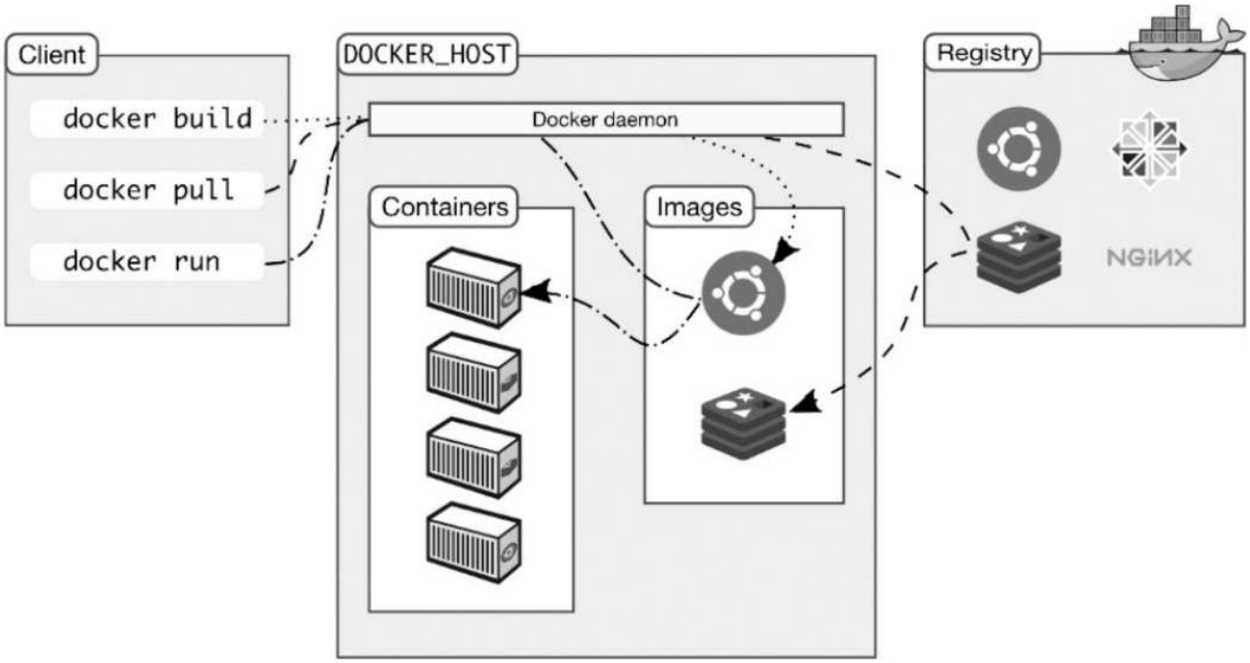


Figure 13-25 *Docker Architecture*

The Docker client is a command-line utility (run with the **docker** command) that talks to the REST API of the Docker daemon so that the administrator can issue commands for the operations and management of Docker containers. The client can communicate to a local or remote Docker instance.

The Docker host is where the Docker daemon resides. The Docker daemon (**dockerd**) is a service that runs on the host operating system and interfaces with the operating system kernel to allow containers to function. The client communicates with the Docker daemon through the REST API. The Docker host houses running containers and also interfaces with the registry to pull container images housed there. When containers are launched, the daemon looks in its local images, and if the appropriate image is there, it launches the container; if the image is not there, the daemon asks the registry for the image file and then stores it locally.

The registry is a place to store container images; it hosts one or more repositories. You can pull images from a registry to run or push new images you create yourself to the registry for storage. Then other hosts in your container environment can make use of your new container image. A registry can be private, for the sole use of a single organization, or public, as in the case of Docker Hub.

Using Docker

The best way to get familiar with Docker is to use it. Download the appropriate version of Docker for your computer and launch its containers. It is available for macOS, Windows, and Linux, using Docker Desktop on macOS or Windows or Docker Engine and CLI tools on Linux.

> **Note**
>
> Docker uses management command grouping (e.g., docker container ls, docker image ls) for clarity. The older flat syntax (docker ps, docker images) still works for backward compatibility, and you'll see it widely in older tutorials and scripts. The addition of the management command syntax adds a hierarchy that will differentiate what you are working on instead of

everything being lumped together. This book uses the management-command syntax. If you were to get a question on the exam that uses the old command line, you could simply omit the first command after docker. For example, the command docker container ps would be shortened to docker ps.

The command **docker** runs the client process, which communicates to the Docker daemon. When you type **docker** and press Enter, you see a long list of available commands. Example 13-1 displays a shortened version of this list.

Example 13-1 *Docker Command List*

```
$ docker

Usage: docker [OPTIONS] COMMAND

A self-sufficient runtime for containers
<cut for brevity>
Management Commands:
  builder     Manage builds
  buildx      Docker Buildx (multi-arch image builds)
  checkpoint  Manage checkpoints
  compose     Docker Compose (multi-container apps)
  config      Manage Docker configs
  container   Manage containers
  context     Manage contexts
  desktop     Docker Desktop
  image       Manage images
  init        Creates Docker-related starter files
  network     Manage networks
  node        Manage Swarm nodes
  plugin      Manage plugins
  scout       Docker Scout (image vulnerability scanning)
  secret      Manage Docker secrets
  service     Manage services
  stack       Manage Docker stacks
  swarm       Manage Swarm
  system      Manage Docker
  trust       Manage trust on Docker images
  volume      Manage volumes
<cut for brevity>

Run 'docker COMMAND --help' for more information on a command.
```

In Example 13-1, notice the list of management commands. These represent the new hierarchy into which Docker groups subcommands. For commands that operate on containers, you can enter **docker container** and then the operation you wish to perform. At any time, you can type **--help** to get more details on what commands are available, as shown in Example 13-2.

Example 13-2 *Docker Container Help*

```
$ docker container --help

Usage: docker container COMMAND

Manage containers

Commands:
  attach      Attach local standard input, output, and error streams to a running
              container
 commit      Create a new image from a container's changes
  cp          Copy files/folders between a container and the local filesystem
  create      Create a new container
  diff        Inspect changes to files or directories on a container's filesystem
  exec        Run a command in a running container
  export      Export a container's filesystem as a tar archive
  inspect     Display detailed information on one or more containers
  kill        Kill one or more running containers
  logs        Fetch the logs of a container
  ls          List containers
  pause       Pause all processes within one or more containers
  port        List port mappings or a specific mapping for the container
  prune       Remove all stopped containers
  rename      Rename a container
  restart     Restart one or more containers
  rm          Remove one or more containers
  run         Run a command in a new container
  start       Start one or more stopped containers
  stats       Display a live stream of container(s) resource usage statistics
  stop        Stop one or more running containers
  top         Display the running processes of a container
  unpause     Unpause all processes within one or more containers
  update      Update configuration of one or more containers
  wait Block until one or more containers stop, then print their exit codes
```

You can run the command **docker container** *COMMAND* **--help** for more information on a command.

For the purposes of the 200-901 CCNAAUTO exam, you are expected to know how to use Docker images in a local developer environment. This means you are not expected to be an expert on all things Docker, but you do need to know how to build, launch, and manage containers on your local machine. The management commands you will use the most for launching and working with Docker are within the **container** and **image** categories. If you are working with a container image, you use the commands under **image**, and if you want to run or stop a container, you use the commands under **container**. As long as you know what you want to work on, mapping it to the command is fairly straightforward.

Working with Containers

When working with containers, the key commands are as follows:

- **create:** Create a container from an image.
- **start:** Start an existing container.
- **run:** Create a new container and start it.
- **ls:** List running containers.
- **inspect:** Get detailed information regarding the container.
- **logs:** Print run logs from the container's execution.
- **stop:** Gracefully stop running the container.
- **kill:** Stop the main process in the container abruptly.
- **rm:** Delete a stopped container.

There are a few ways to launch a container. You may have noticed that the **create**, **start**, and **run** commands seem very similar. The **create** command is used to instantiate a container from an image but does not start the container. Think of it as preloading a container that you can run later. The **start** command starts up the loaded container and allows it to execute. The **run** command, on

the other hand, creates and starts the container as a single command. Example 13-3 provides a quick example of how to launch a sample container with the **run** command.

Example 13-3 *Docker Hello World*

```
$ docker run hello-world
Unable to find image 'hello-world:latest' locally
latest: Pulling from library/hello-world
1b930d010525: Pull complete
Digest: sha256:4df8ca8a7e309c256d60d7971ea14c27672fc0d10c5f303856d7bc48f8cc17ff
Status: Downloaded newer image for hello-world:latest

Hello from Docker!
This message shows that your installation appears to be working correctly.

To generate this message, Docker took the following steps:
 1. The Docker client contacted the Docker daemon.
 2. The Docker daemon pulled the "hello-world" image from the Docker Hub.
 (amd64)
 3. The Docker daemon created a new container from that image which runs the
 executable that produces the output you are currently reading.
 4. The Docker daemon streamed that output to the Docker client, which sent it
 to your terminal.

To try something more ambitious, you can run an Ubuntu container with:
 $ docker run -it ubuntu bash

Share images, automate workflows, and more with a free Docker ID:
 https://hub.docker.com/

For more examples and ideas, visit:
 https://docs.docker.com/get-started/
```

In Example 13-3, Docker looks for the container image hello-world and can't find it in the local image store. Since it has never downloaded it before, it defaults to the Docker Hub registry, does a lookup, finds the image, and then downloads (or pulls, in Docker speak) the latest one. It then starts the container and prints to the console a message saying that the Docker installation is working as expected. Since you now have a container that has been executed, you can use the subcommand **ls** to see what is running:

```
$ docker container ls

CONTAINER ID  IMAGE       COMMAND   CREATED            STATUS

   PORTS   NAMES
```

Unfortunately, you don't see any active containers. That is because the one that was launched ran, printed the message, and then ended. To see containers that have been run but are now stopped, you have to add the **-a** flag to **ls**:

```
$ docker container ls -a

CONTAINER ID  IMAGE        COMMAND   CREATED            STATUS

   PORTS   NAMES

bac0c2a2cca8  hello-world  "/hello"  43 minutes ago     Exited(0) 43 minutes ago
```

Now you can see the container ID, the image name, the command that was issued, and its current status, which in this case is exited. You also see any ports or names that were assigned to the container.

> **Note**
>
> The older flat syntax used ps instead of ls (for example, docker ps -a). You can still type docker container ps -a and get exactly the same output as the previous example, even though the ps command doesn't appear in the menu. This is for legacy compatibility with the pre-1.13 command line.

Launching a container and interacting with it requires a few more commands. Example 13-4 runs a Ubuntu container image (**ubuntu**) in interactive mode and connects the local terminal to the containers terminal and launches the bash shell (**bash**). (The options **-i** and **-t** can be presented separately or together, as **-it**.)

Example 13-4 ***docker run output*** *Command*

```
$ docker container run -it ubuntu bash
Unable to find image 'ubuntu:latest' locally
latest: Pulling from library/ubuntu
7ddbc47eeb70: Pull complete
c1bbdc448b72: Pull complete
8c3b70e39044: Pull complete
45d437916d57: Pull complete
Digest: sha256:6e9f67fa63b0323e9a1e587fd71c561ba48a034504fb804fd26fd8800039835d
Status: Downloaded newer image for ubuntu:latest
root@a583eac3cadb:/#
```

Since this container hasn't been loaded locally before, Docker pulls the image from Docker Hub, layer by layer. It downloads the manifest and layers for the selected Ubuntu image tag. Once it is pulled down, the container is started, and the bash shell is executed. The terminal now shows that you are connected to the container as root (in the container's namespace). If you execute the bash **ps** command, you can see that there are only two things: the bash shell and the **ps** command that you just ran. This highlights the container's isolation from the host OS:

```
root@a583eac3cadb:/# ps

  PID TTY    TIME CMD

    1 pts/0    00:00:00 bash

  11 pts/0    00:00:00 ps
```

To disconnect from the container, you can type **exit**, but that would stop the container. If you want to leave it running, you can hold down Ctrl and press P and then while still holding Ctrl press Q. This sequence drops you back to the host OS and leaves the container running. From the host OS, you can type **docker container ls** to see that your container is actively running:

```
$ docker container ls

CONTAINER ID IMAGE COMMAND CREATED STATUS PORTS NAMES

a583eac3cadb ubuntu "bash" About a minute ago Up About a minute musing_bartik
```

If you want to reconnect to the container, you can use the container ID or the assigned name to attach to the container and interact with it again:

```
$ docker container attach a583eac3cadb

root@a583eac3cadb:/#
```

The **inspect** and **logs** commands are two very useful commands for troubleshooting a container. **inspect** generates a ton of information—pretty much everything having to do with the creation of your container, such as network and runtime environment, mappings, and so on (as shown in Example 13-5). Be warned: Reading the output is like reading a container's DNA.

Example 13-5 *docker container inspect Command*

```
$ docker container inspect a583eac3cadb
[
    {
        "Id":
"a583eac3cadbafca855bec9b57901e1325659f76b37705922db67ebf22fdd925",
        "Created": "2026-04-15T09:35:12.537810374Z",
        "Path": "bash",
        "Args": [],
        "State": {
            "Status": "exited",
            "Running": false,
            "Paused": false,
            "Restarting": false,
            "OOMKilled": false,
            "Dead": false,
            "Pid": 0,
            "ExitCode": 127,
            "Error": "",
            "StartedAt": "2026-04-15T09:35:13.185535918Z",
            "FinishedAt": "2026-04-15T09:53:15.898516767Z"
        },
        "Image":
 "sha256:775349758637aff77bf85e2ff0597e86e3e859183ef0baba8b3e8fc8d3c
        ba51c",
        "ResolvConfPath": "/var/lib/docker/containers/a583eac3cadbafca
855bec9b57901e1325659f76b37705922db67ebf22fdd925/resolv.conf"
<cut for brevity>
```

The **logs** command details everything the container recorded as output and is super helpful for catching error messages:

```
$ docker container logs a583eac3cadb

root@a583eac3cadb:/# ps

   PID TTY    TIME CMD

    1 pts/0    00:00:00 bash

   11 pts/0   00:00:00 ps

root@a583eac3cadb:/# uname

Linux
```

Now that you know the basics on launching a container, the next step is to make it accessible to the rest of the world. To do this, you can start by pulling an nginx web server container and assigning it ports to use with the **-p** flag. This flag requires that you map the container port to a free port on the Docker host. If you use don't map the port explicitly, Docker maps every exposed container port to an ephemeral host port. In addition, you want to make your nginx container stay running in the background instead of just quitting after you launch it. By using the **-d** flag, you can detach the container and leave it running in the background. The last thing you can try is to give the container a name with the **--name** flag. This way, you can refer to it by name instead of by using the random one Docker assigns or the container ID:

```
$ docker container run --name test-nginx -p 80:80 -d nginx

dfe3a47945d2aa1cdc170ebf0220fe8e4784c9287eb84ab0bab7048307b602b9
```

After the container is launched, you can see it running by using the **ls** command, get information on what port it is using (port 80), and see that the container is mapped to the local host IP address (referenced by 0.0.0.0):

```
$ docker container ls
CONTAINER ID IMAGE          COMMAND    CREATED                STATUS  PORTS    NAMES
dfe3a47945d2 nginx "nginx -g 'daemon of..." 21 seconds
ago Up 20 seconds 0.0.0.0:80->80/tcp test-nginx
```

You can also use **docker container port test-nginx** to list just the port mapping.

If you open a web browser and point it to http://localhost, your nginx server should show you a default HTML page (see Figure 13-26).

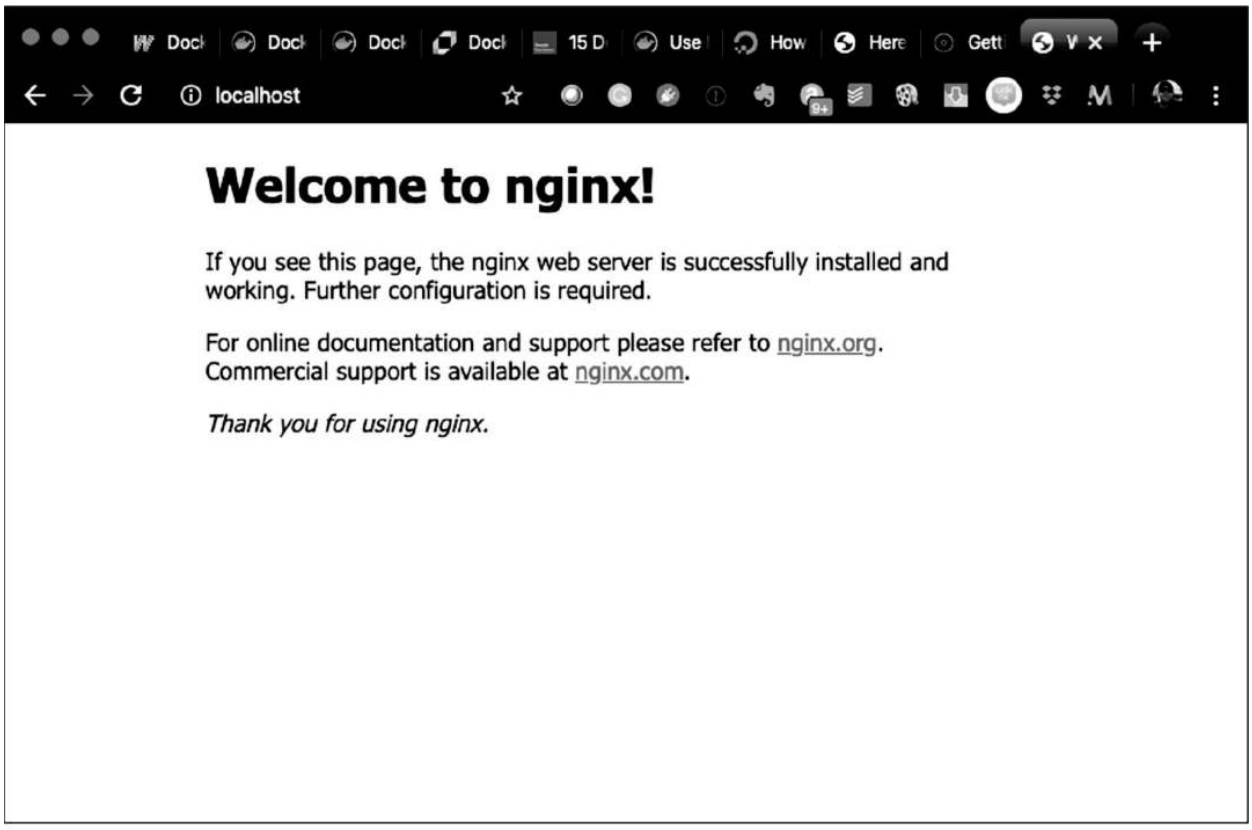


Figure 13-26 *nginx in a Docker Container*

A web server with default content is not very useful. Containers are meant to be read-only, but there are a couple ways to add your own content. There is the **cp** command, which allows you to copy files from your local host to the container when it is up and running; however, in order to save those changes, you would have to store another layer on top of your container and then push that new container to your repository. This would have to be done every time a change is made. A better way is to mount the content directly to the container when it boots up so that it can share the content. This way, if you need to update anything, you just change the files in one directory, and those changes are automatically shared to any container using them. A quick example of this using the nginx server should make this process clearer.

We can start with some HTML files that nginx can display. I have a simple HTML file in a directory called html under Documents on my local laptop running Docker; in macOS, the path is ~/Documents/html. There is a default directory where nginx looks for HTML files to display to users connecting to the service: /usr/share/nginx/html. You have to supply this mapping with the **-v** or **--volume** flag when you launch your container using the previous example. Here is what it looks like:

```
$ docker container run --name test-nginx -p 80:80 -d -v \
~/Documents/html:/usr/share/nginx/html nginx
d0d5c5ac86a2994ea1037bd9005cc8d6bb3970bf998e5867fe392c2f35d8bc1a
```

After you enter this long command, when you connect to http://localhost, you see your own HTML content. The **-v** flag maps your local directory to the shared directory on the container, and from the container's perspective, those files are in /usr/share/nginx/html. Figure 13-27 shows the results of this container mapping to the container host's file system.

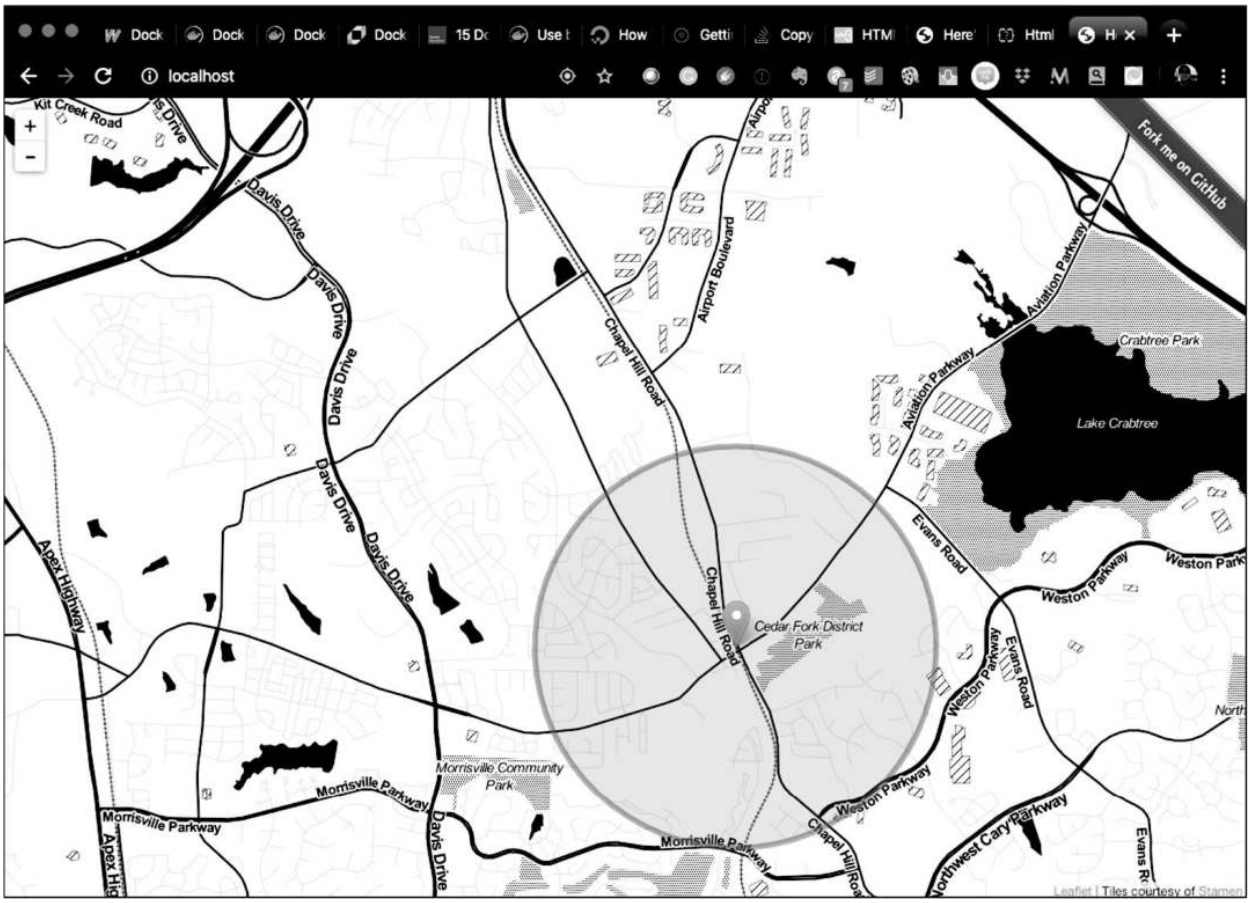


Figure 13-27 *Custom Content Shared from the Local Host*

To stop running a container, you use the **stop** or **kill** command. The difference between the two is pretty obvious. **stop** tries to allow the container time to finish any final work in a graceful way. Using **kill**, on the other hand, is like pulling the plug. For most situations, you want to use **stop**:

```
$ docker container stop test-nginx

test-nginx
```

stop and **kill** both halt the container but don't remove its writable layer and metadata from Docker storage. If you type **docker container ls -a**, you will still see it listed. When you want to remove the container from Docker storage, you can use the **rm** command. You can also use the very handy command **prune** to remove all stopped containers from Docker storage. This is useful on a laptop when you're testing containers and want to free up disk space without having to individually remove multiple containers. Just be careful not to get rid of any container you might actually want to start back up because **prune** will get rid of every inactive container. Here is an example of **prune** in action:

```
$ docker container rm test-nginx

test-nginx



$ docker container prune

WARNING! This will remove all stopped containers.

Are you sure you want to continue? [y/N] y

Deleted Containers:

a583eac3cadbafca855bec9b57901e1325659f76b37705922db67ebf22fdd925
```

Dockerfiles

A Dockerfile is a script that is used to create a container image to your specifications. It is an enormous time saver and provides a ready-made system for replicating and automating container deployments. Once you have built a Dockerfile, you can create an infinite number of the same images without having to manually edit or install software. A Dockerfile is simply a text file with a structured set of commands that Docker executes while building the image. Some of the most common commands are as follows:

- **FROM:** Selects the base image used to start the build process or can be set to **scratch** to build a totally new image.
- **MAINTAINER (deprecated):** Records the image author; use LABEL for author metadata instead.
- **RUN:** Creates image layers and executes commands within a container.
- **CMD:** Executes a single command within a container. Only one can exist in a Dockerfile.
- **WORKDIR:** Sets the path where the command defined with **CMD** is to be executed.
- **ENTRYPOINT:** Executes a default application every time a container is created with the image.
- **ADD:** Copies the files from the local host or remotely via a URL into the container's file system.
- **ENV:** Sets environment variables within the container.
- **EXPOSE:** Documents a port that the application listens on; it does not publish the port.
- **USER:** Sets the UID (or username) of the user that is to run the container.
- **VOLUME:** Sets up a sharable directory that can be mapped to a local host directory.
- **LABEL:** Provides a label to identify the created Docker image.

Creating a Dockerfile starts with creating a text file. The file must be named Dockerfile, and you place it in the working directory where you want to create an image. Once you have opened your favorite editor, the syntax is straightforward. Docker reads the file and operates on it line by line, so the order in which you build your Dockerfile should follow the steps you would perform manually to load your software and configure it. First, you need to select your foundation. This can be any container image, but in this instance, you can use Ubuntu as the foundation. Pin a supported base-image tag instead of relying on latest. You can record the image author with an OCI image label. Here is an example:

```
FROM ubuntu:24.04

LABEL org.opencontainers.image.authors="Cisco Champion <user@domain.com>"
```

Next, you need to specify the software you want to load and what components should be added. Just as when loading software on a Linux server, you first need to run **apt-get update**. For this container, you install nginx and any dependencies:

```
RUN apt-get update && apt-get install -y --no-install-recommends \

    nginx && rm -rf /var/lib/apt/lists/*
```

Now you need to tell Docker what port to expose for networking. In this case, you would expose ports 80 and 443:

```
EXPOSE 80 443
```

Next, you create a sharepoint for HTML files in nginx so that you can give it access to your web files:

```
VOLUME /usr/share/nginx/html
```

Finally, set the container to run nginx on launch:

```
CMD ["nginx", "-g", "daemon off;"]
```

```
FROM ubuntu:24.04

LABEL org.opencontainers.image.authors="Cisco Champion <user@domain.com>"

RUN apt-get update && apt-get install -y --no-install-recommends \

    nginx && rm -rf /var/lib/apt/lists/*

EXPOSE 80 443

VOLUME /usr/share/nginx/html

CMD ["nginx", "-g", "daemon off;"]
```

Now that you have a Dockerfile, it's time to build an image.

Docker Images

> **Note**
>
> Container images are now typically multi-architecture, so the same image tag works on x86_64 and ARM64. That matters because Apple Silicon Macs are ARM64, as are many cloud instance types (AWS Graviton, Azure Cobalt, Google Axion). docker buildx makes building multi-arch images straightforward. For example, docker buildx build --push --platform linux/amd64,linux/arm64 -t myimage . produces both architectures in one push.

When working with Docker images, you primarily use the following commands:

- **build:** Builds an image from a Dockerfile.
- **push:** Pushes a local image to a remote registry for storage and sharing.
- **ls:** Lists images stored locally.
- **history:** Shows the creation and build process for an image.
- **inspect:** Provides detailed information on all aspects of an image, including layer detail.
- **rm:** Deletes an image from local storage.

In the previous example, you built a Dockerfile using the pinned Ubuntu 24.04 base and then updated the package index and installed nginx. You then declared a volume and documented the ports used by the new web server. The next step is to kick off the build process with Docker. While in the directory where you created your Dockerfile, execute the build process as shown in Example 13-6.

Example 13-6 *Building a Container from a Dockerfile*

```
$ docker build -t myimage:latest .
Sending build context to Docker daemon 2.048kB
Step 1/8 : FROM ubuntu:latest
 ---> 775349758637
Step 2/8 : MAINTAINER Cisco Champion (user@domain.com)
 ---> Using cache
 ---> f83e0f07db18
Step 3/8 : RUN apt-get update
 ---> Using cache
 ---> 646cc0e9f256
Step 4/8 : RUN apt-get upgrade -y
 ---> Using cache
 ---> c2701f555b0f
Step 5/8 : RUN apt-get install nginx -y
 ---> Using cache
 ---> 4abf50fd4a02
Step 6/8 : EXPOSE 80 443
 ---> Using cache
 ---> a9a1533064b2
Step 7/8 : VOLUME /usr/share/nginx/html
 ---> Using cache
 ---> 2ecd13c5af2b
Step 8/8 : CMD ["nginx", "-g", "daemon off;"]
 ---> Running in e03bd2387319
Removing intermediate container e03bd2387319
 ---> 04ae8c714993
Successfully built 04ae8c714993
Successfully tagged myimage:latest
```

The build is successful. You can see that with the **-t** flag, you can set a name that can be used to call the image more easily. The **.** at the end of the command tells Docker to look for the Dockerfile in the local directory. You could also use the **-f** flag with a path to the location of Dockerfile file, if it is not in your current directory. Docker also accepts a URL to the Dockerfile, such as at GitHub. As shown in this example, the name **Dockerfile** must be capitalized and one word.

When an image is built, you can use **docker image ls** to see it in local storage:

```
$ docker image ls

REPOSITORY TAG  IMAGE ID     CREATED      SIZE

myimage     latest  04ae8c714993 48 minutes ago 154MB
```

You can use a number of commands to get information from your images. The first is **inspect**. Entering **docker image inspect myimage**, as shown in Example 13-7, gets all the details about the image, including its layer structure.

Example 13-7 *docker image inspect Command*

```
$ docker image inspect myimage
[
    {
        "Id":
"sha256:04ae8c7149937b2ce8b8963d6c34810d8dc53607c9a97064e1f3c85cdc3
        abb46",
        "RepoTags": [
            "myimage:latest"
        ],
        "RepoDigests": [],
        "Parent":
"sha256:2ecd13c5af2b41349b58f2397cbbbc70f5c1c604262113bae443f1f6fc
        3758cc",
        "Comment": "",
        "Created": "2026-04-15T15:53:37.794817007Z",
        "Container": "e03bd238731932ca2cab6c8b7fa1346105b839ce2a8604c0c7d
        34352b907a4af",
        "ContainerConfig": {
            "Hostname": "e03bd2387319",
            "Domainname": "",
            "User": "",
            "AttachStdin": false,
            "AttachStdout": false,
            "AttachStderr": false,
            "ExposedPorts": {
                "443/tcp": {},
                "80/tcp": {}
            },
            "Tty": false,
            "OpenStdin": false,
            "StdinOnce": false,
            "Env": [
                "PATH=/usr/local/sbin:/usr/local/bin:/usr/sbin:/usr/bin:/sbin:/bin"
<Cut for brevity>
```

The **image history** command shows you the creation of the image and all the steps that were taken to build the image, as shown in Example 13-8. This can be very useful in troubleshooting an image build.

Example 13-8 *docker image history Command*

```
$ docker image history myimage
IMAGE          CREATED          CREATED BY                                      SIZE
COMMENT
04ae8c714993   56 minutes ago   /bin/sh -c #(nop)  CMD ["nginx" "-g" "daemon…   0B
2ecd13c5af2b   57 minutes ago   /bin/sh -c #(nop)  VOLUME [/usr/share/nginx/…   0B
a9a1533064b2   57 minutes ago   /bin/sh -c #(nop)  EXPOSE 443 80                0B
4abf50fd4a02   57 minutes ago   /bin/sh -c apt-get install nginx -y             60.2MB
c2701f555b0f   58 minutes ago   /bin/sh -c apt-get upgrade -y                   1.55MB
646cc0e9f256   58 minutes ago   /bin/sh -c apt-get update                       27.6MB
f83e0f07db18   58 minutes ago   /bin/sh -c #(nop)  MAINTAINER Cisco Champion…   0B
775349758637      3 weeks ago   /bin/sh -c #(nop)  CMD ["/bin/bash"]            0B
<missing>         3 weeks ago   /bin/sh -c mkdir -p /run/systemd && echo 'do…   7B
<missing>         3 weeks ago   /bin/sh -c set -xe   && echo '#!/bin/sh' > /…   745B
<missing>         3 weeks ago   /bin/sh -c [ -z "$(apt-get indextargets)" ]     987kB
<missing>         3 weeks ago /bin/sh -c #(nop) ADD file:a48a5dc1b9dbfc632… 63.2MB
```

You probably noticed the missing image IDs at the bottom of the history. The build process uses something called *intermediate layers* to execute commands. These layers are like temporary storage that gets rolled up into the next layer when each stage is finished, and then the intermediate layer is deleted.

When the image is built and ready to go, you can run it by using the following command:

```
$ docker container run -p 80:80 -p 443:443 -d myimage

bf0889f6b27b034427211f105e86cc1bfeae8c3b5ab279ccaf08c114e6794d94


$ docker container ls


CONTAINER ID  IMAGE    COMMAND         CREATED       STATUS

PORTS               NAMES


bf0889f6b27b  myimage "nginx -g        7 seconds ago Up 6 seconds

   0.0.0.0:80->80/tcp,  elastic_maxwell

                      'daemon of..."

0.0.0.0:443->443/tcp
```

You can use the **docker image rm** command to remove an image from storage. This command needs the image ID or a name to select the appropriate image.

Docker Hub

Docker Hub isn't the only registry. GitHub Container Registry (GHCR) is convenient when your code lives on GitHub. Amazon ECR, Google Artifact Registry, and Azure Container Registry are the cloud-native options inside their respective ecosystems. Quay.io (Red Hat) is a popular independent option. The push and pull commands work the same way against any of them. Only the image name's prefix changes (for example, ghcr.io/user/image:tag versus docker.io/user/image:tag).

Docker Hub is a free service that Docker offers to the community to house and share prebuilt container images. It is the world's largest container registry, with millions of images and other artifacts. Chances are that if you are looking to build a container, someone has already put together one that would work for you. Docker Official Images are reviewed and published through Docker's program, but they still require normal vulnerability scanning and update practices. Docker Personal accounts include unlimited public repositories and one private repository; paid plans raise limits and add features. Docker Hub is a cloud-only service, so if you want to run a private container registry in your data center, you can deploy a private registry implementation on premises.

Docker Hub has a slick web interface that makes it easy to set up a repository and secure it. It also integrates with GitHub and Bitbucket to enable automated container creation from source code stored in your version control system. It's a great way to start building a CI/CD pipeline as it is free to get started. If you set up a free account, you can take advantage of Docker Hub's capabilities immediately. Figure 13-28 shows Docker Hub signup page.

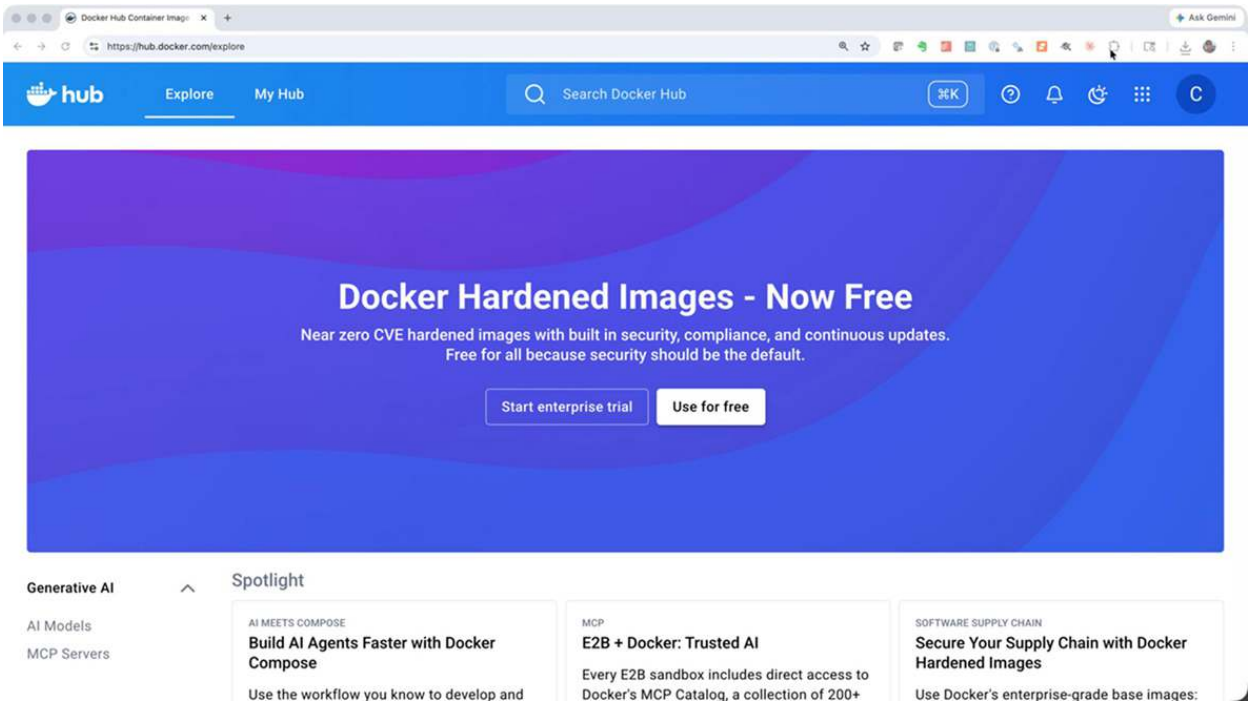


Figure 13-28 *Signing Up for Docker Hub*

Once you have an account, you can search through the many containers available and can filter on operating system, category, and whether or not the image is an official Docker certified image, verified by the publisher, or an official image published by Docker itself. Figure 13-29 shows the image search function.

Docker Desktop ships with a graphical Dashboard for interacting with containers on your local host as well as Docker Hub. This view can download and launch containers with a simple point and click. It's great for people who are struggling with all of the command-line options that the typical Docker client requires. (Older content may reference Kitematic, which Docker archived in 2021.) But be warned: The Docker Desktop Dashboard gives you a quick visual interface, though the docker CLI still gives you the most control. Open Docker Desktop and explore the Containers, Images, and Volumes views. Figure 13-30 shows the Docker Desktop Dashboard.

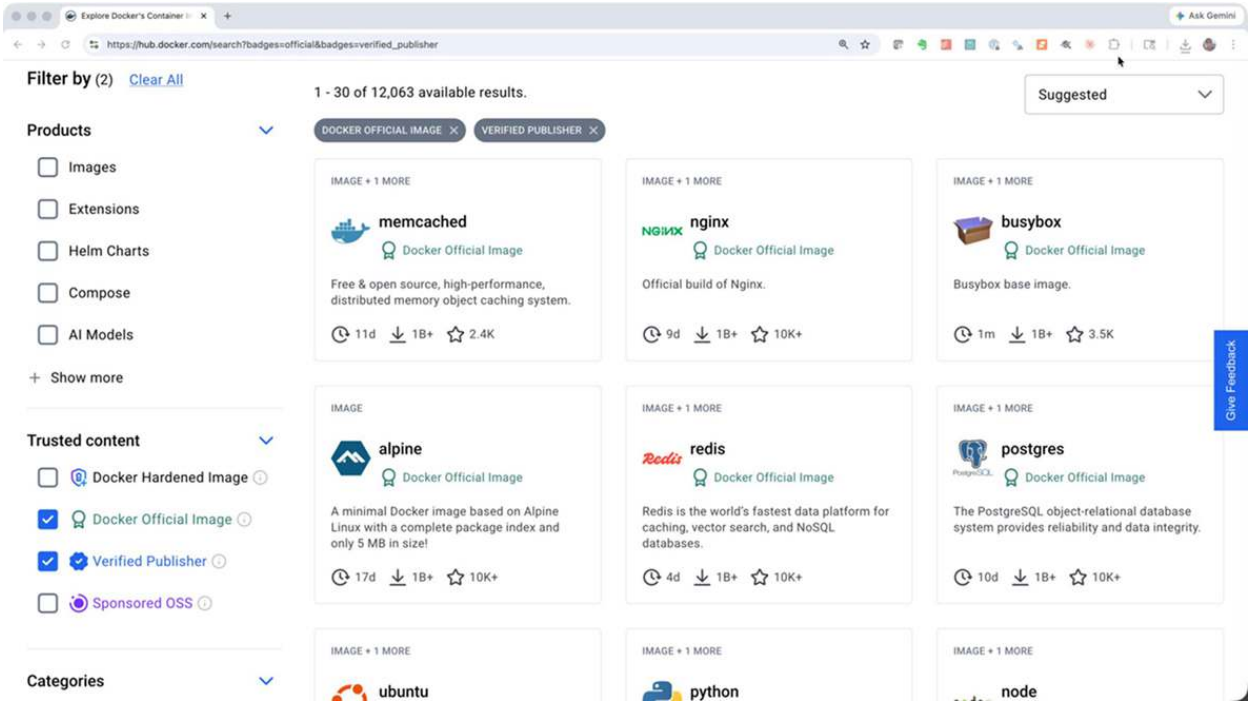


Figure 13-29 *Docker Hub Image Search*

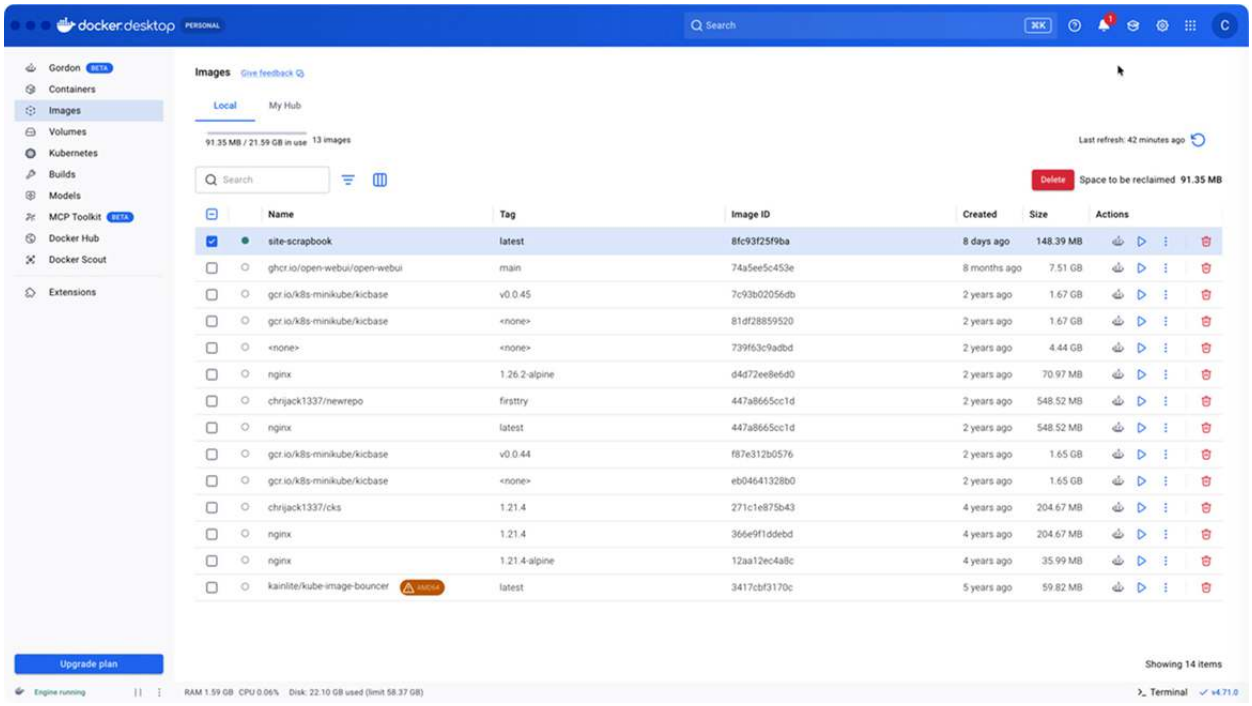


Figure 13-30 *Docker Desktop Dashboard*

After you build a container, as you did in the previous example, you need to push it to a repository. Since you can use Docker Hub for free, you can set up a private repository to store this container. If you wanted to share with the world the container you created, you could do so by simply making the repository public. A private repository is always a good bet for code that is just for your organization. Figure 13-31 shows how to set up a private repo in Docker Hub named newrepo that you can push your new container to.

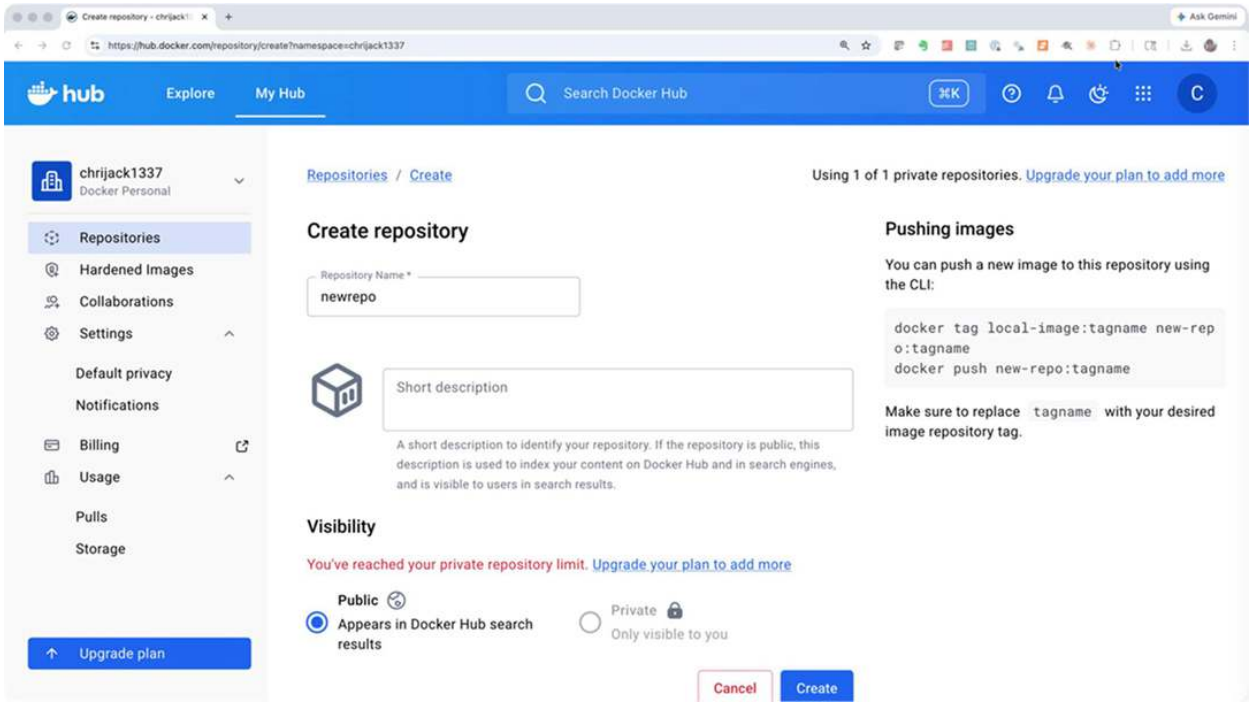


Figure 13-31 *Docker Hub Repository Setup*

Once your repository is set up, you need to tag your image to be pushed to Docker Hub. Use **docker image ls** to list your images and take note of the image ID:

```
$ docker image ls

REPOSITORY     TAG        IMAGE ID       CREATED              SIZE

myimage        latest     04ae8c714993   About an hour ago    154MB
```

Next, you need to issue the **docker image tag** command with the image ID of your image and assign it to *username*/newrepo:firsttry. newrepo simply identifies the repo you want to place the image in, and the tag after the : allows you to differentiate this particular image from any other in the repository (in this instance, it is named firsttry):

```
$ docker image tag 04ae8c714993 chrijack1337/newrepo:firsttry
```

Now you are ready to issue the **push** command, as shown in Example 13-9, and you see Docker start to push the layers of your container image to the repository.

Example 13-9 ***docker image push*** *Command*

```
$ docker image push chrijack1337/newrepo:firsttry
The push refers to repository [docker.io/chrijack1337/newrepo]
3e92d80e0ac4: Pushed
a87bf84680fc: Pushed
02d0765ebf97: Pushed
e0b3afb09dc3: Pushed
6c01b5a53aac: Pushed
2c6ac8e5063e: Pushed
cc967c529ced: Pushed
firsttry: digest: sha256:01cf95854003cd1312fb09286d41bc6bfd9fe3fb82f63e66e5060c7fd5a6230a size:
1786
```

You can now check back with Docker Hub and see that the image is now hosted in your private repository (see Figure 13-32).

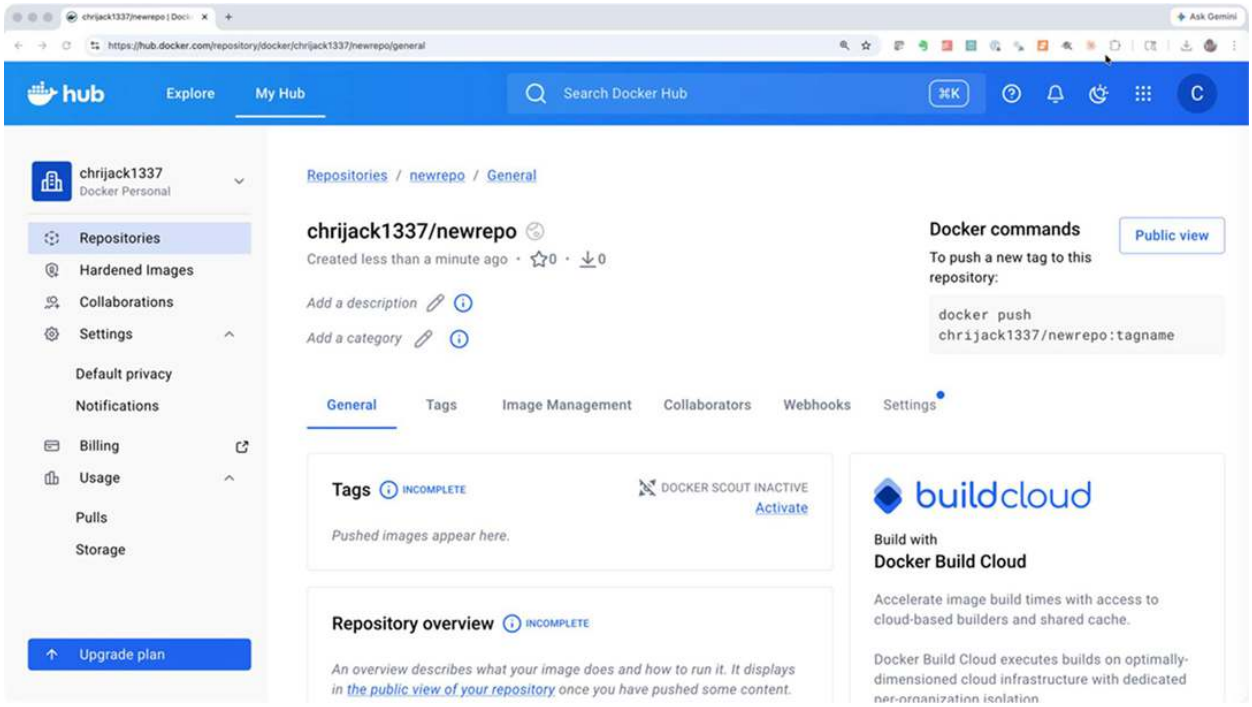


Figure 13-32 *Docker Hub Pushed Image*

After this, any time you want to retrieve the image, you can type **docker image pull** *username*/**newrepo:firsttry**, and Docker will load it in your local image storage.

There is quite a bit more that you can do with Docker, but this section should get you started and focused on what you need to know for the 200-901 CCNAAUTO exam. Make sure that to review the Docker documentation and try these examples on your own computer. Nothing beats practical experience.

Application Security

Modern application deployment exposes a much larger surface area than the monoliths of a decade ago. Containers, microservices, APIs, and cloud-managed databases each carry their own security concerns. Three areas matter for any deployed application: protecting secrets, encrypting data, and handling that data responsibly.

Secret Protection

A secret is any piece of information that grants access to something: API keys, database passwords, TLS private keys, OAuth client secrets, or signing keys. A common application security mistake is putting secrets in source code or in container images.

Secrets do not belong in:

- Source code (a developer commits the file; the secret is now in Git history forever).
- Configuration files committed to source control.
- Container images (anyone who can pull the image can extract the file).
- Environment variables baked into Dockerfiles.

Where secrets do belong:

- A dedicated secrets manager such as HashiCorp Vault, AWS Secrets Manager, Azure Key Vault, or Google Secret Manager.
- Kubernetes Secrets (with encryption at rest enabled and RBAC restricting access).
- Docker secrets for Docker Swarm deployments.

The application reads its secrets at runtime from these stores rather than carrying them along inside the image or repo.

Encryption

Encryption protects data in two states: at rest (sitting on disk or in a database) and in transit (moving over a network).

Encryption in transit is commonly provided by TLS. Every client-to-server call should use HTTPS or another TLS-protected protocol. Modern infrastructure makes this easy: Let's Encrypt provides free certificates, cloud load balancers can terminate TLS, and service meshes can be configured to encrypt internal traffic between microservices.

Encryption at rest happens at multiple layers. Disk-level encryption (LUKS on Linux, BitLocker on Windows, transparent encryption on AWS EBS or Azure Disks) protects the entire storage volume. Database-level encryption protects specific data even if the disk is mounted. Application-level encryption gives you the most control. You encrypt sensitive fields before storing them, and it's essential for highly sensitive data such as Social Security numbers or health records.

Data Handling

How you handle data is as important as how you encrypt it. A few principles apply to any deployed application:

- Collect only the minimum data you actually need. Every additional field is a potential breach disclosure.
- Know what's personally identifiable. Names, emails, IP addresses, device identifiers, and biometrics are all PII in most jurisdictions.
- Define a retention policy and stick to it. Old data still in the database is still your problem in a breach.
- Securely delete data when its purpose ends. Soft-deletes (a "deleted" flag in a row) are not deletion.
- Log access to sensitive data. You can't investigate something you didn't log.

> **Note**
>
> Container images can carry vulnerabilities from outdated base images, OS packages, or pulled-in libraries. Image scanning tools (Trivy, Docker Scout in Docker Desktop, Snyk Container, Grype) analyze your image layers and report known CVEs before you deploy. Run scans in your CI pipeline so vulnerabilities get caught at build time rather than discovered in production.

Network Services in Application Deployment

A deployed application doesn't sit alone. It sits behind and beside several network services that together make it reachable, scalable, and safe. The four you'll see in essentially every web application deployment are firewalls, DNS, load balancers, and reverse proxies.

Firewall

A firewall enforces a policy about which network traffic is allowed. In application deployment, firewalls operate at multiple levels.

Perimeter firewalls (or cloud security groups in AWS, Azure, and GCP) sit between the public internet and your application. They typically allow inbound traffic only on the ports your application actually needs (usually 443 for HTTPS) and block everything else. East-west firewalls or microsegmentation policies control traffic between services inside your own infrastructure, preventing

a compromised front-end from reaching the database directly, for example. Container platforms have their own equivalent. When supported by the cluster's network plugin, Kubernetes Network Policies define which pods can talk to which.

DNS

DNS resolves human-readable names into IP addresses. In an application deployment context, DNS does double duty.

External DNS publishes your application's name (api.example.com) to the public internet, pointing at your load balancer's IP. Internal DNS handles service discovery inside your environment, where your microservices look up each other by name (orders.internal, payments.internal) rather than by hardcoded IP. In a Kubernetes cluster, the built-in DNS service handles this automatically; within the same namespace, you can reach a service by its short name and the cluster takes care of routing.

Load Balancer

A load balancer distributes incoming traffic across multiple instances of your application so that no single instance is overwhelmed and a single failure does not take you down.

Layer 4 (transport) load balancers route based on IP and port: fast and protocol-agnostic. Layer 7 (application) load balancers can inspect HTTP headers, paths, and cookies to make routing decisions, enabling smart features like routing /api requests to one backend and /admin to another. Health checks let the load balancer detect a dead instance and stop sending traffic to it. Session affinity (or sticky sessions) binds a user to a specific backend when application state requires it. Common load balancers include AWS Application Load Balancer, NGINX, HAProxy, and Envoy.

Reverse Proxy

A reverse proxy sits between clients and your application, hiding the internal topology and adding a layer of services.

A reverse proxy commonly handles TLS termination (centralizing client-facing certificate management; backends may still use TLS or mTLS), caching (serving repeated requests from memory), request routing (sending paths to the right backend), and request rewriting. NGINX, HAProxy, Traefik, and Envoy are all common reverse proxies. Often the same software acts as both load balancer and reverse proxy. It may also be the application's edge security layer (a Web Application Firewall).

How They Compose

In a typical deployment, traffic flows like this:

A user hits api.example.com. DNS resolves it to the load balancer's public IP. The firewall allows the connection on port 443. The load balancer terminates TLS and routes to a healthy backend. A reverse proxy in front of the backend adds caching and routing. The request finally reaches your application.

Top OWASP Threats

The Open Worldwide Application Security Project (OWASP) publishes the OWASP Top 10, a regularly updated list of the most critical web application security risks. The current edition is OWASP Top 10:2025. As an application developer or someone deploying applications, you don't need to memorize every entry, but you do need to recognize the categories and know how to think about them.

The OWASP Top 10:2025 categories are:

- A01:2025—Broken Access Control: Authorization is missing or implemented incorrectly, letting users access data or actions they shouldn't. CSRF is included in this category.
- A02:2025—Security Misconfiguration: Default credentials, unnecessary features enabled, missing security headers, or unprotected admin interfaces.
- A03:2025—Software Supply Chain Failures: Compromised dependencies, malicious packages, build pipeline tampering.
- A04:2025—Cryptographic Failures: Sensitive data left unencrypted, weak ciphers, leaked keys.
- A05:2025—Injection: User input that gets interpreted as code or commands. Includes SQL injection, command injection, and cross-site scripting (XSS).
- A06:2025—Insecure Design: Architectural flaws that enable attacks even if the code is implemented correctly.
- A07:2025—Authentication Failures: Weak password policies, missing multi-factor authentication, broken session management.
- A08:2025—Software or Data Integrity Failures: Trusting unverified updates, unsigned packages, or tampered data.
- A09:2025—Security Logging and Alerting Failures: Not knowing you're under attack until well after the breach.
- A10:2025—Mishandling of Exceptional Conditions: Errors that fail open, leak stack traces, or skip security checks.

Injection: SQL Injection and XSS

Injection attacks happen when an attacker's input gets interpreted as code rather than as data. SQL injection and cross-site scripting (XSS) are the two most familiar examples.

SQL injection: A web form takes a username and the application builds a query like SELECT * FROM users WHERE name = '<input>'. If the input is admin' OR '1'='1, the query becomes SELECT * FROM users WHERE name = 'admin' OR '1'='1', which returns every user. The fix is parameterized queries (also called prepared statements). The database driver treats user input as data rather than as part of the SQL syntax.

Cross-site scripting (XSS): An attacker submits content (a comment, a profile field) that contains <script>...</script>, and the application renders it back to other users without escaping it. The script runs in the victim's browser with the victim's session. The fix is output encoding: escape every value you render based on its context (HTML, attribute, JavaScript). Modern templating frameworks (Jinja, React, Vue) escape by default.

CSRF

Cross-site request forgery (CSRF) tricks a logged-in user's browser into submitting a request to a site they're authenticated to without the user knowing.

For example, a user is logged into bank.example.com in one tab and visits a malicious page in another tab. The malicious page contains a hidden form that auto-submits a fund transfer to bank.example.com. The user's browser sends the cookies along, the bank trusts the session, and the transfer goes through. Modern defenses include anti-CSRF tokens (a per-request secret only the legitimate page knows), the SameSite cookie attribute (which can restrict cookies on cross-site requests), and custom request headers for API endpoints. Most modern web frameworks (Django, Rails, Spring, ASP.NET) include CSRF protection out of the box.

Where to Learn More

OWASP publishes detailed guidance for each category, complete with code examples and prevention checklists, at owasp.org/Top10/. They also publish the OWASP Cheat Sheet Series, short focused guides on specific topics like authentication, secrets management, and session management. Both live at cheatsheetseries.owasp.org. Both are free, community-maintained, and an essential resource for anyone deploying applications.

Exam Preparation Tasks

As mentioned in the section "How to Use This Book" in the Introduction, you have a couple of choices for exam preparation: the exercises here, Chapter 19, "Final Preparation," and the exam simulation questions on the companion website.

Review All Key Topics

Review the most important topics in this chapter, noted with the Key Topic icon in the outer margin of the page. Table 13-2 lists these key topics and the page number on which each is found.

Table 13-2 *Key Topics*

Key Topic Element	Description	Page Number
Paragraph	NIST definitions of cloud	
Paragraph	Cloud deployment models	
Paragraph	Edge computing model	
Paragraph	Application deployment methods	
Paragraph	The three ways of DevOps	
Paragraph	Implementing DevOps	
Paragraph	Docker architecture	
Note	Command line change	
Paragraph	Dockerfiles	
List	Docker images	
Paragraph	Docker Hub	

Define Key Terms

Define the following key terms from this chapter and check your answers in the glossary:

software as a service (SaaS)
private cloud
public cloud
hybrid cloud
edge computing
container
DevOps
registry
artifact repository
Docker
Docker daemon
continuous integration/continuous deployment (CI/CD)
Docker Hub

Additional Resources

Periodic Table of DevSecOps Tools: **https://digital.ai/learn/devsecops-periodic-table/**

Docker CLI Cheat Sheet: **https://docs.docker.com/get-started/docker_cheatsheet.pdf**

Docker Basics: How to Use Dockerfiles: https://thenewstack.io/docker-basics-how-to-use-dockerfiles/

Chapter 14. Application Security

This chapter covers the following topics:

- **Identifying Potential Risks:** This section introduces some of the concepts involved in application security and shows how to identify potential risks in applications.
- **Protecting Applications:** This section shows how to protect an application from various vulnerabilities using various frameworks and leveraging firewalls, IDSs, and IPSs.

Application security involves making applications more secure by proactively finding, fixing, and enhancing their safety. Much of what happens during the development phase includes tools and methods to protect apps once deployed. Today, application security is paramount. Many tools have emerged to secure networks, storage, and even code over the years. Various coding tools and methodologies can be used to identify and mitigate inadvertent code threats.

This chapter will teach you about application security issues and how applications can be secured using modern networking components. This chapter also provides an overview of the Open Web Application Security Project (OWASP) and what it brings to the table for application developers.

"Do I Know This Already?" Quiz

The "Do I Know This Already?" quiz allows you to assess whether you should read this entire chapter thoroughly or jump to the "Exam Preparation Tasks" section. If you are in doubt about your answers to these questions or your own assessment of your knowledge of the topics, read the entire chapter. Table 14-1 lists the major headings in this chapter and their corresponding "Do I Know This Already?" quiz questions. You can find the answers in Appendix A, "Answers to the 'Do I Know This Already?' Quiz Questions."

Table 14-1 *"Do I Know This Already?" Section-to-Question Mapping*

Foundation Topics Section	Questions
Identifying Potential Risks	1–4
Protecting Applications	5–8

Caution

The goal of self-assessment is to gauge your mastery of the topics in this chapter. If you do not know the answer to a question or are only partially sure of the answer, you should mark that question as wrong for purposes of self-assessment. Giving yourself credit for an answer that you correctly guess skews your self-assessment results and might provide you with a false sense of security.

1. A vulnerability is a _____ in protection efforts. It can be exploited by threats to gain unauthorized access to an asset.
   - **a.** strength
   - **b.** weakness
   - **c.** neutral condition
   - **d.** side effect
2. Which of the following are threats? (Choose two.)
   - **a.** Phishing
   - **b.** Man-in-the-middle (on-path-attack)
   - **c.** JWT token

d. Brute force

3. What type of test is used to identify the possible weak spots in applications, servers, or networks?

   a. Pen

   b. White box

   c. Eraser

   d. Unit

4. Which of the following is a tool that can be used for network discovery and security auditing?

   a. Nslookup

   b. Nmap

   c. ps

   d. curl

5. What is the minimum number of identity components MFA uses to authenticate a user's identity?

   a. One

   b. Two

   c. Three

   d. Four

6. Which of the following is used for fingerprinting of data to detect whether the data has been modified?

   a. Private key

   b. Public key

   c. One-way hash

   d. Certificate

7. Data needs to be secured in multiple locations. Which of the following are the correct locations?

   a. Memory, storage, and network

   b. Flash, wire, and optics

   c. Hard disk, database, and analytics

   d. AWS, Google Cloud, and Azure

8. Which modes best describe IDSs and IPSs, respectively?

   a. Passive, passive

   b. Passive, active

   c. Active, passive

   d. Active, active

Foundation Topics

Identifying Potential Risks

The National Institute of Standards and Technology (NIST) defines a framework called the Cybersecurity Framework, which is shown in Figure 14-1. The framework organizes necessary cybersecurity activities into the following functions:

- **Identify:** An organization needs to understand what kinds of cybersecurity risks can affect daily operations. Areas such as data theft, network breaches, and employee information are some of the risks that need to be identified.
- **Protect:** An organization needs to understand what it can do to prevent attacks. Protection could include deploying proper networking elements such as firewalls and tools for better software development. Protection helps minimize the impact of any attack.
- **Detect:** It is important to install tools that detect data breaches or attacks in a time-sensitive manner, while attacks are happening.
- **Respond:** An organization needs to have a plan to deal with an attack. It needs to know what procedures must be followed and what can be done to minimize the impact of an attack.
- **Recover:** An organization needs to quickly resolve any services or systems that have been affected.

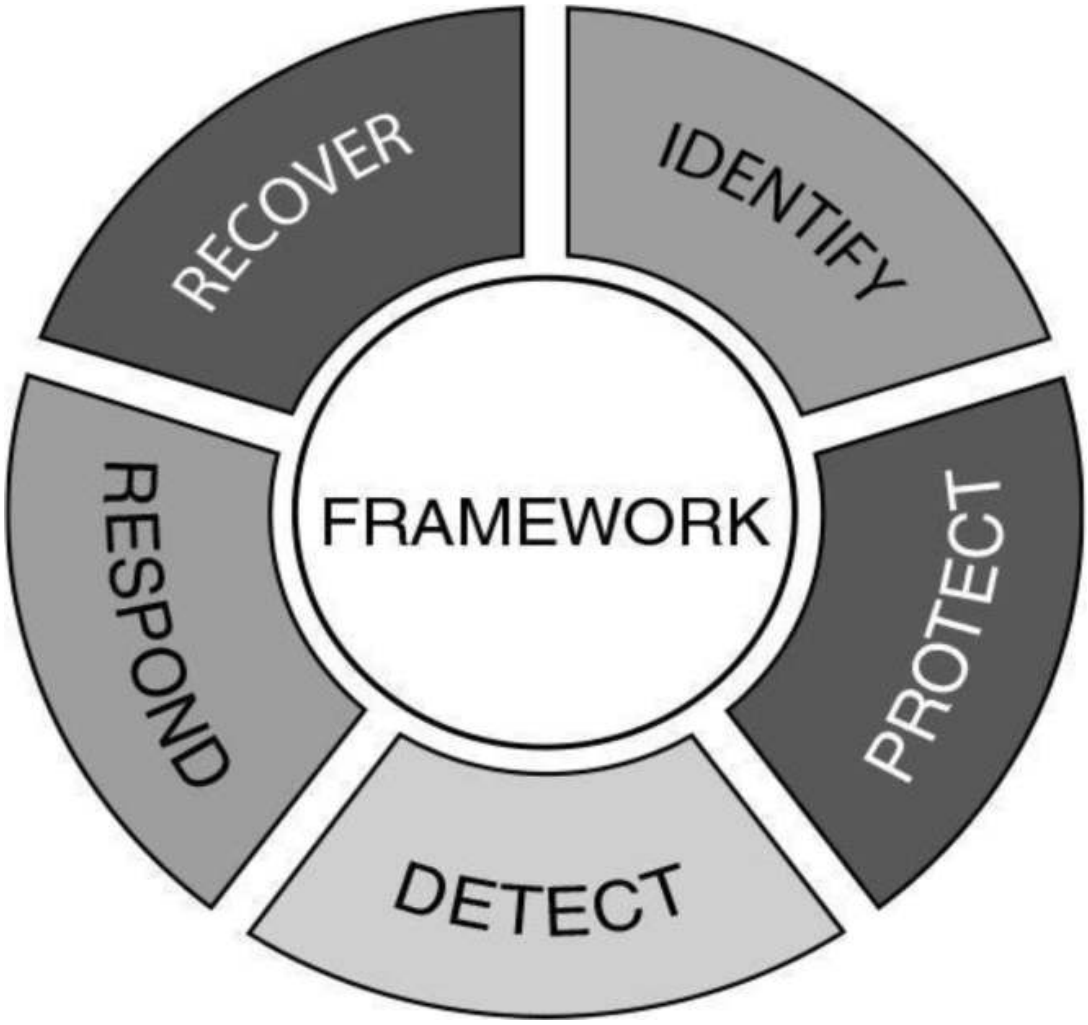


Figure 14-1 *NIST Cybersecurity Framework*

For more details about the Cybersecurity Framework, visit https://www.nist.gov/cyberframework/online-learning/components-framework.

This chapter explains application security using the Cybersecurity Framework as a substratum to understand some key concepts. To ensure application security, it is important to identify threats and what they affect.

Before we talk about the potential risks, it is essential to understand some key terms and their relationships. The following are some commonly misunderstood terms:

- **Asset:** An asset is something you're trying to protect. It could be information, data, people, or physical property. Assets are of two types: tangible and intangible. Information and data may include databases, software code, and critical company records. People include employees and customers. Intangible assets include reputation and proprietary information.
- **Threat:** Anything you are trying to protect against is known as a threat. A threat is any code or person that can exploit a vulnerability when it has access to an asset and can control or damage the asset.
- **Vulnerability:** A vulnerability is a weakness or gap in protection efforts that threats can exploit to gain unauthorized access to an asset.
- **Risk:** Risk is the intersection of assets, threats, and vulnerabilities. Certain risks can potentially compensate for an asset's loss, damage, or destruction due to a threat exploiting a vulnerability. Threats can always exist, but there is little or no risk if vulnerabilities don't exist. Similarly, you can have a weakness but little or no risk if you have no threat.

Common Threats and Mitigations

In this section, we examine the most common threats. Table 14-2 describes some of the dangers to application and some mitigation tips for each one.

Table 14-2 *Threats and Sample Mitigation Options*

Threat	What It Does	Mitigation Options
Buffer overflow	An attacker uses a program to try to cause an application to store that input in a buffer that isn't large enough. The attacker's data overwrites memory beyond the buffer's allocated space. Attackers can use a buffer overflow to modify a computer's memory and undermine or take control of program execution.	• Separate executable memory from non-executable memory. • Randomize address spaces for data. • Use the built-in protection options in newer software OSs and languages.
Man-in-the-middle (On-path-attack)	Attackers insert themselves between two endpoints (a browser and a web server) and intercept or modify communications between them. The attackers can then collect information and impersonate either of the two agents. In addition to targeting websites, these attacks can target email communications, DNS lookups, and public Wi-Fi networks.	• Adopt a Secure Sockets Layer (SSL)/Transport Layer Security (TLS) strategy for both web and email. • Avoid sensitive data in public Wi-Fi or computers.
Denial-of-service (DoS) attack	A threat actor sends multiple requests that flood the server or networks with traffic to exhaust resources and bandwidth. As system performance degrades, the system becomes increasingly nonresponsive, and legitimate requests are left unfulfilled. These attacks can be coordinated, with multiple devices launching attacks simultaneously. Such an attack is known as a distributed	• Use specially designed protection services (cloud or network).

	denial-of-service (DDoS) attack.	
Cross-site scripting (XSS)	An attacker attaches code to a legitimate website that executes when the victim loads it. This is the typical process: 1. The webpage is loaded, and the malicious code copies the user's cookies. 2. The system sends an HTTP request to an attacker's web server with the stolen cookies in the request's body. 3. The attacker can then use cookies to access sensitive data.	• Validate and sanitize input data. • Employ cookie security, such as timeouts, encoding the client IP address, etc.
Phishing	A threat actor procures sensitive information—, typically usernames, passwords, and so on—, from emails or web pages.	• Educate users to avoid falling for the bait. • Detect and mark emails and sites as spam.
Malware	Malware is a piece of malicious code such as spyware, ransomware, a virus, or a worm. Malware is usually triggered when someone clicks a link or an email attachment and inadvertently installs malicious software.	• Deploy technologies that continually monitor and detect malware evading perimeter defenses.

SQL injection	Structured Query Language (SQL) injection is a code injection technique used to modify or retrieve data from SQL databases. An attacker can execute commands allowing data retrieval from the database by inserting specialized SQL statements into an entry field.	• Use character escaping. • Use stored procedures as opposed to queries. Enforce privileges.
Brute force	A threat actor may use trial and error to decode data. Brute-force methods can be used to crack passwords and crack encryption keys. Other targets include API keys, SSH logins, and Wi-Fi passwords.	• Lock the system after a specified number of attempts. Use two-factor authorization.

Open Web Application Security Project

The goal of a penetration (pen) test is to identify possible weak links in applications, servers, or networks that could be used by an attacker to gain sensitive information or privileged access. It is important to detect such vulnerabilities not only to know that they exist and calculate the risk attached to them but also to try to mitigate them or reduce them to the minimum risk level.

In an era of cybercrime, hackers exploit software vulnerabilities daily. To address security threats, software development teams must prioritize security. The OWASP Foundation's OWASP Top 10 lists critical security risks in modern web applications. The Open Web Application Security Project (OWASP) is a community that produces and articulates various reports, tools, methodologies, and technologies for web application security. OWASP classifies and defines many application security vulnerabilities.

An updated version has been released, and the following section will outline the changes since the 2017 publication.

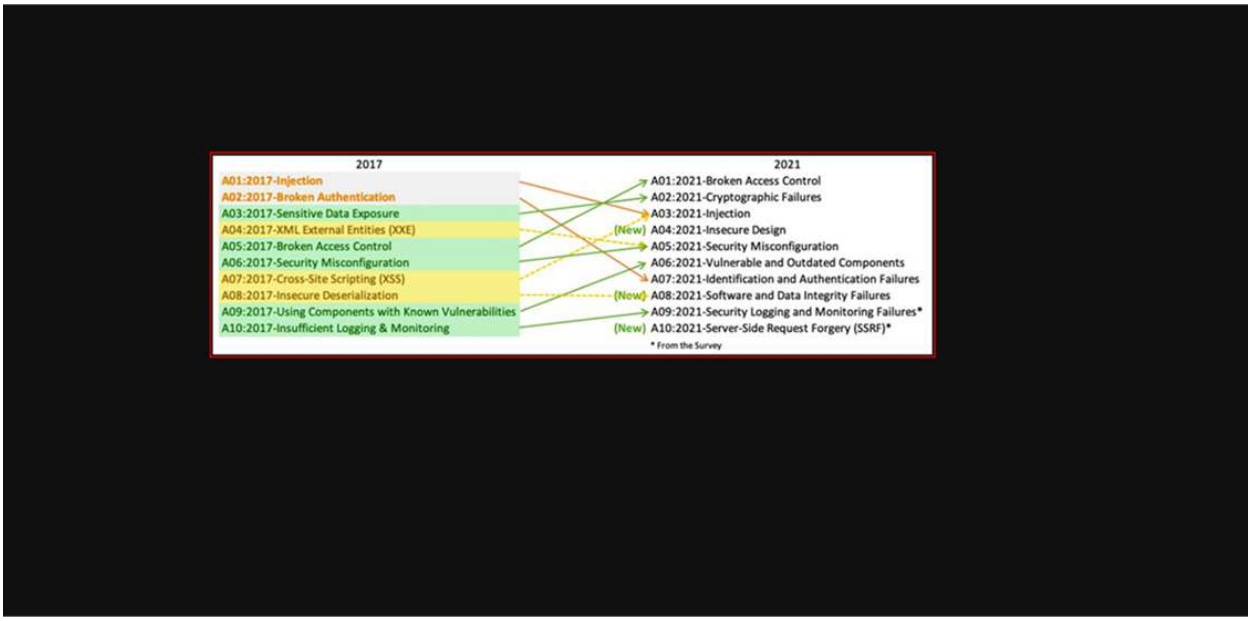


Figure 14-2 *New categories and consolidation in the Top Ten*

Here's a concise summary of the OWASP Top 10, which highlights critical security risks for web applications:

1. **Broken Access Control:** Flaws that allow unauthorized access to sensitive data or functionality.
   Example: A user can manipulate the URL to access another user's private profile without proper authorization.
   Mitigation: Implement strict access controls based on roles and permissions.
2. **Cryptographic Failures:** Issues related to encryption and key management.
   Example: Storing sensitive data (like passwords) in plain text instead of using proper encryption.
   Mitigation: Use strong encryption algorithms and secure key management practices.
3. **Injection**: Vulnerabilities that allow malicious code execution (e.g., SQL injection).
   Example: An attacker injects malicious SQL code into an input field, gaining unauthorized access to the database.
   Mitigation: Validate and sanitize user input to prevent code injection.
4. **Insecure Design**: Risks stemming from poor system architecture and design choices.
   Example: A system lacks proper segregation of duties, allowing a single user to perform critical actions.
   Mitigation: Design systems with security in mind from the start.
5. **Security Misconfiguration:** Errors in configuring security settings (e.g., default passwords).
   Example: Leaving default credentials (e.g., admin/admin) unchanged in a web application.
   Mitigation: Regularly review and update security settings.
6. **Vulnerable and Outdated Components:** Risks associated with using outdated libraries or frameworks.
   Example: Using an outdated library with known security flaws.
   Mitigation: Keep components up to date and monitor for vulnerabilities.
7. **Identification and Authentication Failures:** Problems with user authentication and session management.
   Example: Weak password policies or session management flaws.
   Mitigation: Implement strong authentication mechanisms and manage sessions securely.
8. **Software Quality Issues:** Bugs, logic flaws, and other code-related issues.
   Example: Unhandled exceptions leading to system crashes or exposing sensitive data.
   Mitigation: Conduct thorough code reviews and testing.
9. **Sensitive Data Exposure:** Mishandling of sensitive information (e.g., credit card data).
   Example: Storing credit card details insecurely, leading to data leaks.
   Mitigation: Encrypt sensitive data at rest and in transit.
10. **Broken Function-Level Authorization**: Flaws in authorization checks for specific functions.
    Example: A user can access administrative functions without proper authorization.
    Mitigation: Validate authorization for each function or endpoint.

The CVE (Common Vulnerabilities and Exposures) lists publicly disclosed computer security vulnerabilities. When someone refers to a CVE, they usually mean the CVE ID number assigned to a security defect. Security advisories issued by vendors and researchers almost always mention at least one CVE ID. CVEs help IT professionals coordinate their efforts to prioritize and address these vulnerabilities to make computer systems more secure. CVEs are supervised by the MITRE Corporation, with funding from the Cybersecurity and Infrastructure Security Agency, part of the U.S. Department of Homeland Security. A CVE record usually provides a short one-line description. Details are usually available on sites such as the U.S. National Vulnerability Database NVD (https://nvd.nist.gov/) and the CERT/CC Vulnerability Notes Database (https://www.kb.cert.org/vuls/), as well as the sites of prominent vendors in the industry.

A CVE record consists of the following:

- ID
- Description
- Impact (low/moderate/important/critical)
- Date published

A CVE record would look something like this:

- **ID:** CVE-2020-5313

- **Description:** An out-of-bounds read was discovered in python-pillow in the way it decodes FLI images. An application that uses python-pillow to load untrusted images may be vulnerable to this flaw, allowing attackers to read the application's memory that they should not be allowed to read.
- **Impact:** Moderate
- **Date:** January 3, 2020

Using Nmap for Vulnerability Scanning

Nmap, short for Network Mapper, is a free open-source utility for network discovery and security auditing. Nmap is commonly used for network scanning or vulnerability scanning. Target users for this tool are pen testers, security professionals, and system administrators. Nmap provides detailed, real-time snapshot information about the devices or hosts on a network. Nmap primarily provides three functionalities.

- It provides detailed information on every active IP address on a network, and each IP address can then be scanned for more details if needed.
- It provides a list of live hosts and open ports and identifies the OS of every connected device. This makes Nmap an excellent system-monitoring and pen-testing tool.
- It helps identify security vulnerabilities to protect against attacks.

The best way to get familiar with Nmap is to use it. Nmap is available on macOS and Linux by default. Example 14-1 shows some of the command-line options available.

Example 14-1 *Using Nmap for Network and Vulnerability Scanning*

```
$ nmap --help

Nmap 7.80 ( https://nmap.org )

Usage: nmap [Scan Type(s)] [Options] {target specification}

TARGET SPECIFICATION:

  Can pass hostnames, IP addresses, networks, etc.

  Ex: scanme.nmap.org, microsoft.com/24, 192.168.0.1; 10.0.0-255.1-254

  -iL <inputfilename>: Input from list of hosts/networks

  -iR <num hosts>: Choose random targets

  --exclude <host1[,host2][,host3],...>: Exclude hosts/networks

  --excludefile <exclude_file>: Exclude list from file

HOST DISCOVERY:

  -sL: List Scan - simply list targets to scan

  -sn: Ping Scan - disable port scan

  -Pn: Treat all hosts as online -- skip host discovery

  -PS/PA/PU/PY[portlist]: TCP SYN/ACK, UDP or SCTP discovery to given ports

  -PE/PP/PM: ICMP echo, timestamp, and netmask request discovery probes

  -PO[protocol list]: IP Protocol Ping

  -n/-R: Never do DNS resolution/Always resolve [default: sometimes]

  --dns-servers <serv1[,serv2],...>: Specify custom DNS servers

  --system-dns: Use OS's DNS resolver

  --traceroute: Trace hop path to each host


<cut for brevity>
```

Basic Nmap Scan Against an IP Address or a Host

Many switch options can be used with Nmap, and here we focus on a practical one. To run Nmap against an IP address or a host, you can scan the hostname with the **nmap** *hostname* command, as shown in Example 14-2. (In this example, *hostname* is www.google.com, but you can replace it with any IP address or hostname, including localhost.) Use the **-vv** option as shown in this example to see a more verbose output.

Example 14-2 *Using Nmap to Get Details About a Host or an IP Address*

```
$ nmap -vv www.google.com

Starting Nmap 7.80 ( https://nmap.org ) at 2019-12-08 22:10 PST

Warning: Hostname www.google.com resolves to 2 IPs. Using 216.58.194.196.

Initiating Ping Scan at 22:10

Scanning www.google.com (216.58.194.196) [2 ports]

Completed Ping Scan at 22:10, 0.02s elapsed (1 total hosts)

Initiating Parallel DNS resolution of 1 host. at 22:10

Completed Parallel DNS resolution of 1 host. at 22:10, 0.06s elapsed

Initiating Connect Scan at 22:10

Scanning www.google.com (216.58.194.196) [1000 ports]

Discovered open port 443/tcp on 216.58.194.196

Discovered open port 80/tcp on 216.58.194.196

Completed Connect Scan at 22:10, 8.49s elapsed (1000 total ports)

Nmap scan report for www.google.com (216.58.194.196)

Host is up, received syn-ack (0.025s latency).

Other addresses for www.google.com (not scanned): 2607:f8b0:4005:804::2004

rDNS record for 216.58.194.196: sfo03s01-in-f196.1e100.net

Scanned at 2019-12-08 22:10:05 PST for 9s

Not shown: 998 filtered ports

Reason: 998 no-responses

PORT    STATE SERVICE REASON

80/tcp  open  http     syn-ack

443/tcp open  https    syn-ack
```

CVE Detection Using Nmap

One of Nmap's most magnificent features for finding vulnerabilities is called the Nmap Scripting Engine (NSE). The NSE allows you to use a predefined script or write your own using Lua programming language.

Using NSE is a crucial part of automating system and vulnerability scans. It requires the following syntax:

```
nmap -Pn --script vuln hostname
```

Example 14-3 shows a vulnerability scan for a device on my home network.

Example 14-3 *Using the Nmap Scripting Engine*

```
$ nmap -Pn --script vuln 10.168.243.179

Starting Nmap 7.80 ( https://nmap.org ) at 2019-12-08 22:18 PST

Pre-scan script results:

| broadcast-avahi-dos:

|   Discovered hosts:

|     224.0.0.251

|   After NULL UDP avahi packet DoS (CVE-2011-1002).

|_  Hosts are all up (not vulnerable).

Illegal character(s) in hostname -- replacing with '*'

Nmap scan report for RX-V677*B9772F.hsd1.ca.domain.net (10.168.243.179)

Host is up (0.029s latency).

Not shown: 995 closed ports

PORT      STATE SERVICE

80/tcp    open  http

|_clamav-exec: ERROR: Script execution failed (use -d to debug)

| http-csrf:

| Spidering limited to: maxdepth=3; maxpagecount=20; withinhost=RX-

V677*B9772F.hsd1.ca.domain.net

|   Found the following possible CSRF vulnerabilities:

|

|     Path: http://RX-V677*B9772F.hsd1.ca.domain.net:80/

|     Form id: recoveryform

|_    Form action: /Config/avr_recovery.cgi

|_http-dombased-xss: Couldn't find any DOM based XSS.

| http-fileupload-exploiter:

|

|     Couldn't find a file-type field.

|

|     Couldn't find a file-type field.

|

|     Couldn't find a file-type field.

|

|     Couldn't find a file-type field.

|

|     Couldn't find a file-type field.

|
```

```
|     Couldn't find a file-type field.
|
|     Couldn't find a file-type field.
|
|     Failed to upload and execute a payload.
|
|     Failed to upload and execute a payload.
|
|     Failed to upload and execute a payload.
|
|     Failed to upload and execute a payload.
|
|_    Failed to upload and execute a payload.
|_http-stored-xss: Couldn't find any stored XSS vulnerabilities.
1029/tcp  open  ms-lsa
|_clamav-exec: ERROR: Script execution failed (use -d to debug)
1900/tcp  open  upnp
|_clamav-exec: ERROR: Script execution failed (use -d to debug)
8080/tcp  open  http-proxy
|_clamav-exec: ERROR: Script execution failed (use -d to debug)
|_http-aspnet-debug: ERROR: Script execution failed (use -d to debug)
| http-enum:
|_  /test.html: Test page
| http-slowloris-check:
|   VULNERABLE:
|   Slowloris DOS attack
|     State: LIKELY VULNERABLE
|     IDs:  CVE:CVE-2007-6750
|       Slowloris tries to keep many connections to the target web server
open and
        hold
|       them open as long as possible.  It accomplishes this by opening
connections
        to
|       the target web server and sending a partial request. By doing so, it
starves
```

```
|       the http server's resources causing Denial Of Service.
|
|     Disclosure date: 2009-09-17
|     References:
|       https://cve.mitre.org/cgi-bin/cvename.cgi?name=CVE-2007-6750
|_      http://ha.ckers.org/slowloris/
50000/tcp open  ibm-db2
|_clamav-exec: ERROR: Script execution failed (use -d to debug)


Nmap done: 1 IP address (1 host up) scanned in 130.29 seconds
```

Protecting Applications

An important step in protecting applications is to recognize the risks. Before we talk about the potential risks, it is essential to understand some key terms and their relationships:

- **Hacker or attacker:** These terms refer to people who seek to gain by exploiting weaknesses in software and computer systems. Most of the time, hackers have benign intentions and are simply motivated by curiosity; however, their actions may violate the intended use of the systems they are exploiting. The results can range from mere mischief (such as creating a virus with no intentionally negative impact) to malicious activity (such as stealing or altering information).
- **Malicious code:** Malicious code is unwanted files or programs that can cause harm to a computer or compromise data stored on a computer. Malicious code includes viruses, worms, and Trojan horses.

Tiers of Securing and Protecting

Enterprises depend on applications to run their businesses. Apps not only help customers and partners connect with the enterprise, but often, they also enable employees to get their jobs done. Unfortunately, applications remain one of the most commonly exploited threat vectors. An enterprise must secure and protect web, mobile, and API applications from being compromised and prevent data breaches. As mentioned earlier, application security is a framework that involves making applications more secure, and therefore, an end-to-end approach is needed.

The multilayered software architecture is one of the most famous architectural patterns. Figure 14-3 shows a simple three-tier application architecture that has the following foundations:

- **Tier 1 (Presentation):** This tier presents content to the end user through a web user interface, a mobile app, or via APIs. To present the content, this tier must interact with the other tiers. From a security standpoint, it is very important that access be authorized, timed, and encrypted and that the attack surface be minimized.
- **Tier 2 (Application):** This is the middle tier of the architecture, where the application's business logic runs. The components of this tier typically run on one or more application servers; hence, from a security standpoint, load balancing, limiting access, and proxying help.
- **Tier 3 (Data):** This is the lowest tier of this architecture, mainly concerned with storing and retrieving application data. Application data is typically stored in a database server, a file server, or another device or medium that supports data access logic and provides the necessary steps to ensure that only the data is exposed, without providing access to the data storage and retrieval mechanisms.

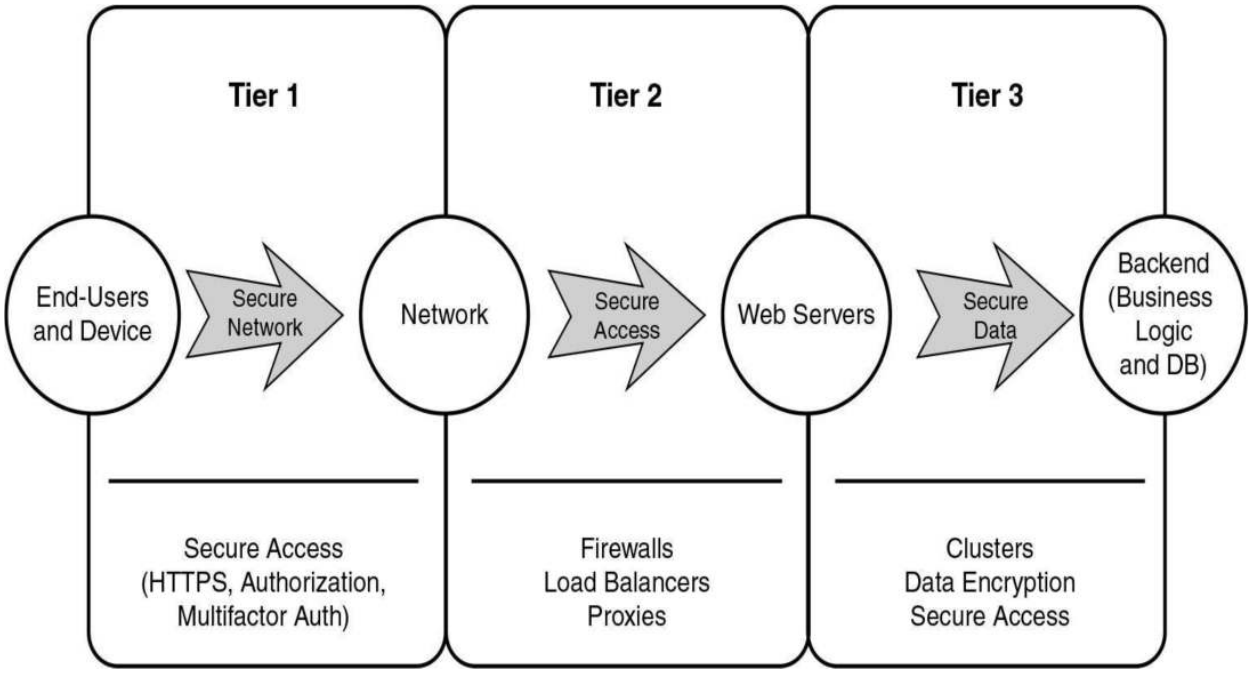


Figure 14-3 *Three-Tier Approach to Application Security*

To minimize risks, the following are some of the best practices in the industry:

- **Keep software up-to-date:** Install software patches so that attackers cannot exploit known problems or vulnerabilities. Many operating systems offer automatic updates.
- **Install end-user or device security:** Install endpoint security software on an end-user's device to keep viruses and malware as far away as possible.
- **Use strong passwords:** A strong password ensures only authorized users can access resources. With strong passwords, it becomes hard to guess and hence decreases security risk.
- **Implement multifactor authentication (MFA):** Authentication validates a user's identity. Attackers commonly exploit weak authentication processes. MFA uses at least two identity components to authenticate a user's identity, minimizing the risk of a cyberattacker gaining access to an account by knowing the username and password.
- **Install a firewall.** Firewalls may be able to prevent some types of attack vectors by blocking malicious traffic before it can enter a computer system and restricting unnecessary outbound communications.
- **Encrypt data:** Ensure that data cannot be accessed even if storage can be reached.

Encryption Fundamentals

Cryptography is the science of transmitting information securely against potential third-party adversaries. The main objectives of cryptography are the following:

- **Confidentiality:** Guaranteeing that the information exchanged between two parties is confidential between them and is not visible to anyone else
- **Integrity:** Ensuring that the integrity of a message is not changed while the message is in transit
- **Availability:** Ensuring that systems are available to fulfill requests all the time

Encryption is the operation that applies an encryption key to plaintext using an encryption algorithm. It turns the plaintext into ciphertext. Decryption is the inverse operation: The decryption key is applied to the ciphertext, and the result is the original plaintext. Both encryption and decryption involve symmetric keys or public/private key pairs.

Public Key Encryption

Public key encryption is a method of encrypting data that involves a pair of keys known as a public key and a private key (or a public/private key pair). The public key is usually published, and the corresponding private key is kept secret. Data that is encrypted with the public key can be decrypted only with the corresponding private key. Figure 14-4 illustrates a simplified way to understand public key encryption.

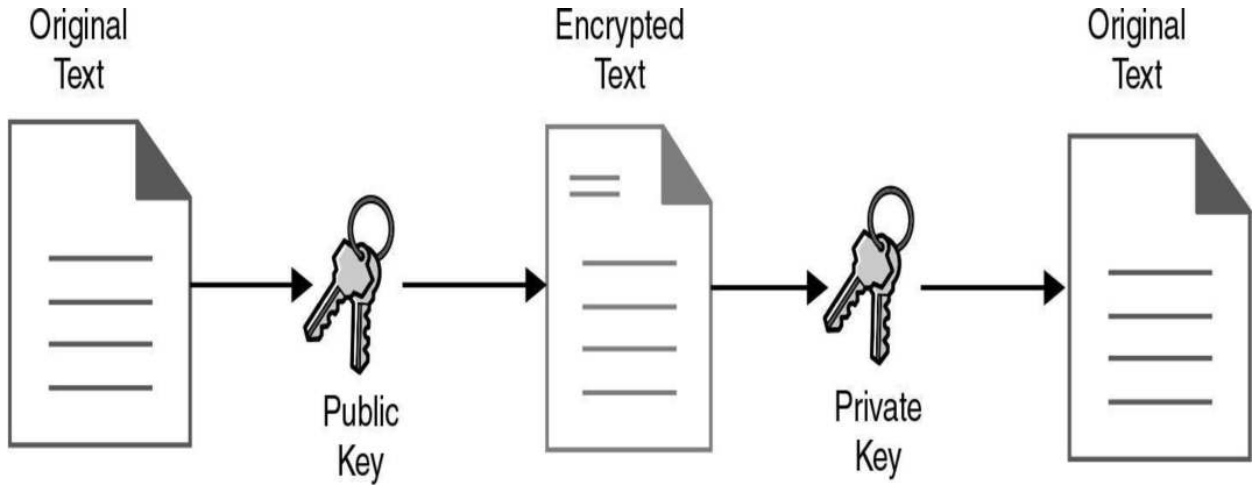


Figure 14-4 *Simple Public Key Encryption*

Public key encryption is used to establish secure communications over the Internet (via HTTPS). A website's SSL/TLS certificate, which is shared publicly, contains the public key, and the private key is installed on the web server.

A TLS handshake uses a public key to authenticate the origin server's identity and exchange data used for generating the session keys. A key exchange algorithm, such as Rivest–Shamir–Adleman (RSA), uses the public/private key pair to agree on session keys, which are used for symmetric encryption when the handshake is complete. Clients and servers can agree on new session keys for each communication session so that bad actors cannot decrypt communications even if they identify or steal one of the session keys.

Data Integrity (One-Way Hash)

As data and documents move across networks or storage devices, ensuring that this data is identically maintained during any operation is crucial.

In the case of data transfer, data flowing between applications in a public network environment can flow across many network elements, each of which can "see" the data. Encryption ensures that even though these elements can see the data, they cannot understand it. However, because the data can flow across nodes you do not control, there is a risk that a node in the network could alter the data before it reaches the destination.

You cannot prevent the data from being altered by someone on the network, so the receiving element (destination) must be able to detect whether data has been modified and, if so, not pass the corrupted data to the application. A one-way hash is used for this purpose. The message hash is a fingerprint of the data. If the data changes, the fingerprint (the message digest or hash) also changes. Someone who alters the data would have no idea what the corresponding digest will be for the modified data. The content of the hashed data cannot be determined from the hash. This is why it is called a *one-way* hash.

Digital Signatures

You can use a private key for encryption and your public key for decryption. Rather than encrypting the data itself, you can create a one-way hash of the data and then use the private key to encrypt the hash. The encrypted hash and other information, such as the hashing algorithm, are known as digital signatures. Figure 14-5 illustrates using a digital signature to validate the integrity of signed data. The data and the digital signature are sent across the network. On the receiving end, two hashes are calculated: one from the received data and the other by decrypting the digital signature using the public key. If the two hashes match, you know that the private and public keys match and the sender's identity is verified.

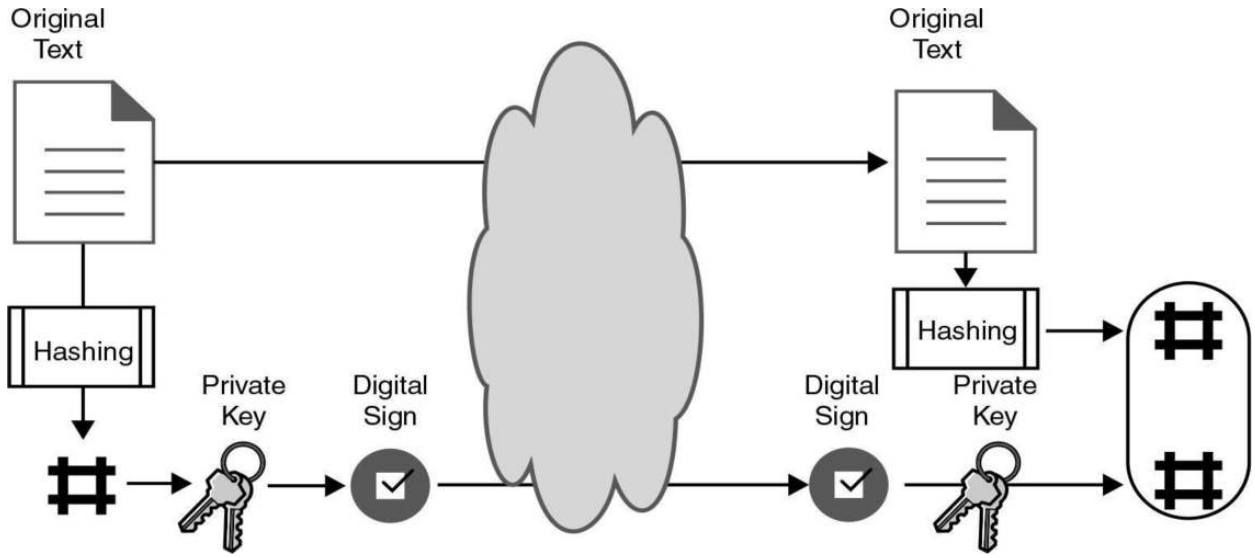


Figure 14-5 *Digitally Signed Data*

Data Security

Today, all our interactions with the world involve online systems in some way, and all these systems deal with data. Data is typically an organization's most valuable asset and is very often sensitive. Data security is the process of securing data and protecting it from unauthorized access. As data traverses various interaction points, it must be secured at all these places. Data needs to be protected in three places, as shown in Figure 14-6:

- Network (data in motion)
- Storage (data at rest)
- Memory (data in use)

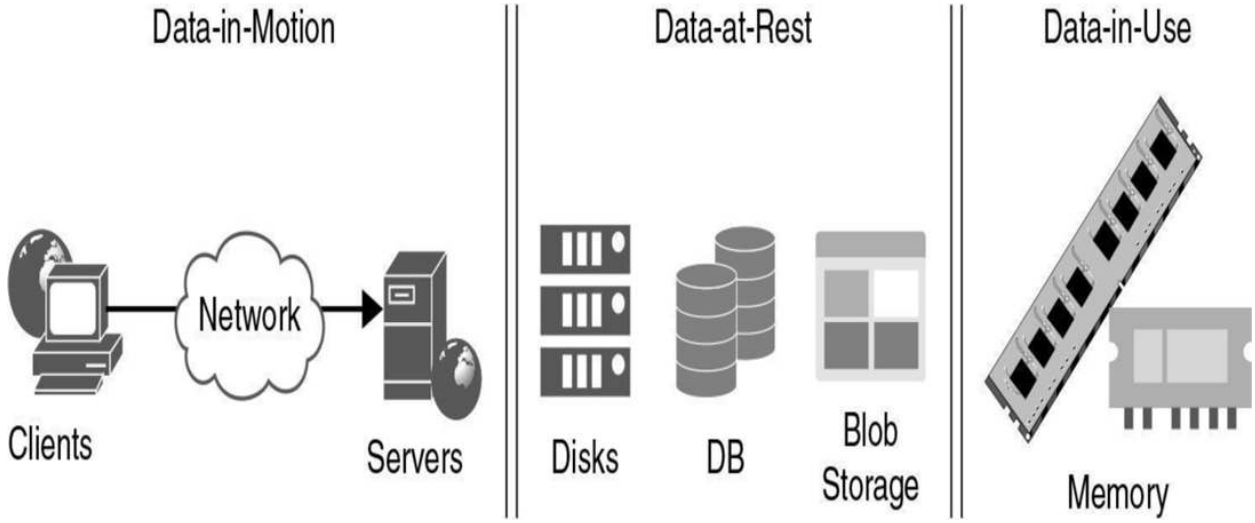


Figure 14-6 *Data Security*

Table 14-3 shows each type of security and describes some ways to secure or protect each type.

Table 14-3 *Data Security Types*

Data Type	Description	Examples
Data in motion (network)	Data travels across various networks and entities. It is prone to be inspected, and data can be stolen. Transport Layer Security (TLS) is a widely adopted security protocol designed to facilitate privacy and data security for communication over the Internet. A primary use case of TLS is encrypting the communication between web applications and servers, such as web browsers loading a website.	• HTTPS • Firewalls
Data at rest (storage)	Any data that is stored can be accessed digitally or physically. Encrypting data at rest protects it against physical theft of the file system storage devices, protects against unauthorized access to data, and satisfies information security or regulatory requirements.	• Full disk encryption • File system encryption • Database encryption
Data in use (memory)	When a running application is processing data, the data is in memory (or in use). Operating systems consider all open files data in use. When an encrypted file is unencrypted in memory, it is vulnerable to being stolen or "scraped."	• Full memory encryption • CPU-based key storage (keys stored in CPU registers) • Enclaves, which are guarded and secure memory segments

Secure Development Methods

As mentioned earlier in this chapter, the application security process starts during development. Instead of trying to bring in security at the end of the development process, secure development needs to be baked in from the start. Addressing security issues from the beginning saves a company time and money in the long run.

It is a common practice for corporations to use some type of software development lifecycle (SDLC). Best practices in secure software development suggest integrating security aspects into each phase of the SDLC. Figure 14-7 show the various aspects of the SDLC.

As you can see in the figure, the SDLC includes these steps:

- **Training:** Training helps get everyone on the project teams into a security frame of mind. Train developers on the team to analyze the business application attack surface and the associated potential threats. Not just developers but all team members should understand the exposure points of their applications (user inputs, front-facing code, exposed function calls, and so on) and take steps to design more secure systems wherever possible.

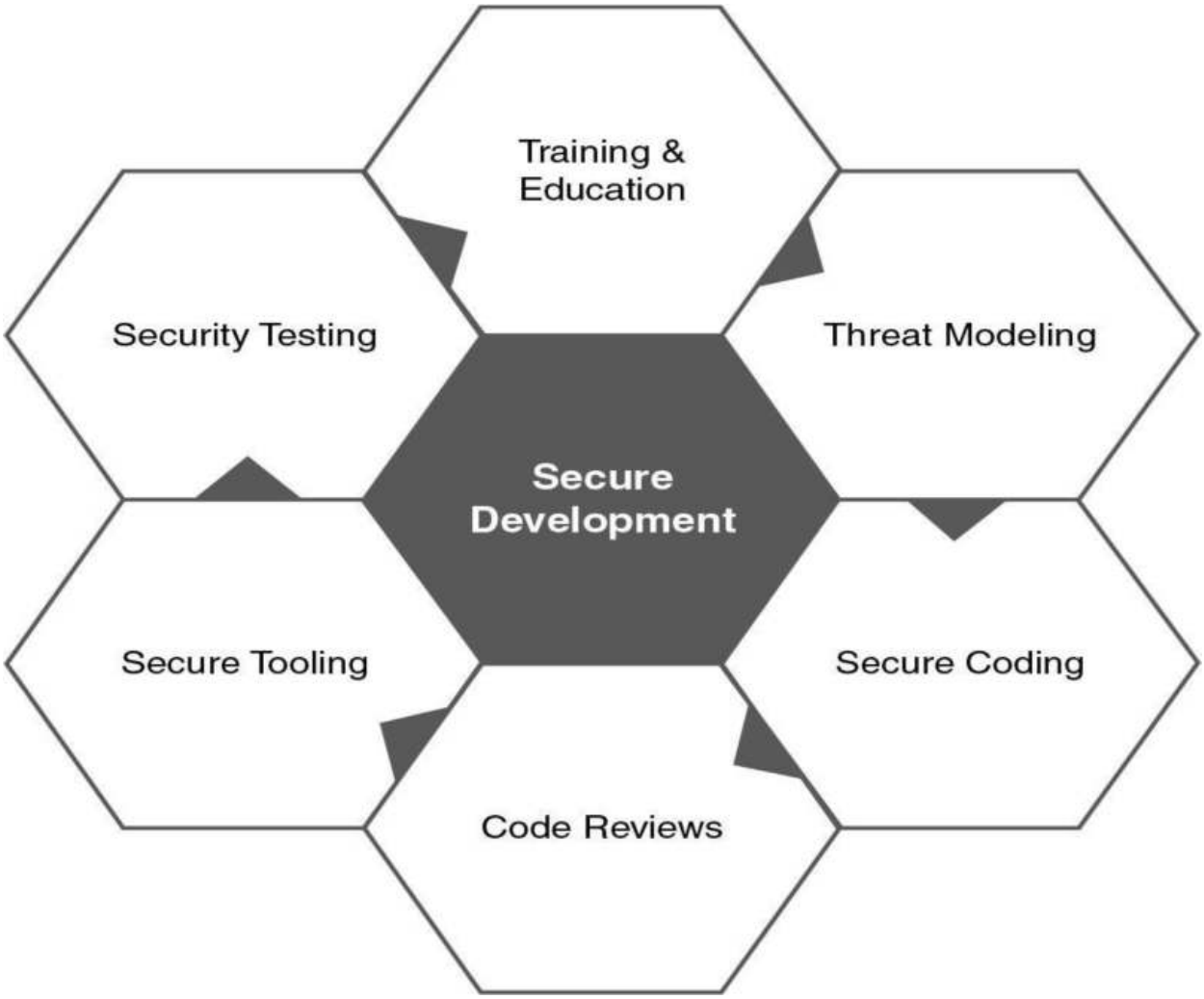


Figure 14-7 *Secure Development*

- **Threat modeling:** For every component and module in a system, ask the "what-how-what" questions: What can go wrong? How can someone try to hack into it? What can we do to prevent this from happening? Various frameworks for threat modeling are available, including the following:
  - STRIDE (Spoofing, Tampering, Repudiation, Information Leak, DoS, Elevation of Privilege)
  - PASTA (Process for Attack Simulation and Threat Analysis)
  - VAST (Visual, Agile, and Simple Threat Modeling)
- **Secure coding:** Build and use code libraries that are already secure or approved by an official committee. The industry has several guidelines for secure coding, and we have listed some of the standard ones here:
  - Validating inputs
  - Encoding output
  - Ensuring authentication and credential management
  - Managing sessions
  - Using access control lists
  - Monitoring error handling and logging
  - Protecting data, including files, databases, and memory
- **Code review:** Code review is one of the most essential steps in securing an application. Usually, the rule that you have to keep in mind is that pen testing and other forms of testing should not be discovering new vulnerabilities. Code review has to be a way to make sure that an application is self-defending. It should also be conducted using a combination of tools and human effort. It is important to designate a security lead who can help review the code from a security point of view.
- **Secure tooling:** Static analysis helps catch vulnerabilities. Its tools detect errors or potential errors in a program's structure and can be useful for documentation or understanding a program. Static analysis is a cost-effective way to discover errors.

Data flow analysis is a form of static analysis that concentrates on programs' use of data and detects some data flow anomalies.

- Dlint (see https://github.com/duo-labs/dlint) is a Duo Labs (Cisco) tool that defines and checks for common best practices when writing secure Python. To evaluate various rules over a code base, Dlint leverages Flake8. Flake8 does the heavy lifting of parsing Python's AST, allowing you to focus on writing robust rule sets.
- **Testing:** Testing includes penetration (pen) testing and system testing, as well as black box and white box testing. Black box testing is a method used to test software without knowing the code's or program's internal structure. Testing teams usually do this testing, and programming knowledge is not required. Black box testing includes functional, behavioral, and closed box testing. On the other hand, white box testing is a software testing method in which the internal structure is known to the tester. Generally, this type of testing is carried out by software developers. Programming knowledge is usually required for white box testing. White box testing includes structural testing, logic testing, path testing, loop testing, code coverage testing, and open box testing.

Testing involves the following steps:

- **Intelligence gathering:** Define the goals, understand what's included in the application, and identify potential areas of vulnerabilities.
- **Scanning:** Understand both running and non-running behaviors. Static analysis tools enable developers and testers to see faults without running an application. These tools can save a lot of time and effort in the long run, and the more errors and defects found here, the better. Dynamic analysis, on the other hand, involves running the application in a real or virtual environment. Usually, many external services and interactions are exercised here.
- **Access:** Use various methods to hack the application, such as testing it for SQL injection, back doors, traffic interception, and so on. Long-term access testing looks at the vulnerabilities exposed when a system is exploited for a long time.
- **Reporting:** Include all details on the vulnerabilities and sensitive data exposed and how long the system remained unhacked.

Securing Network Devices

Network devices are the components of a network that transport communications needed for data, applications, services, and multimedia. These devices include routers, firewalls, switches, servers, load balancers, intrusion detection systems, Domain Name System servers, and storage area networks. Because most or all organizational and customer traffic must pass through them, these devices are ideal targets for malicious actors.

Firewalls

A firewall is a barrier that's put in place to limit damage. A firewall can be hardware or software installed to limit damage from external and internal cyberattacks. It monitors and controls network traffic based on a set of security rules. Figure 14-8 shows how firewalls can monitor and control network traffic as it flows between a local-area network (LAN) and the Internet (WAN).

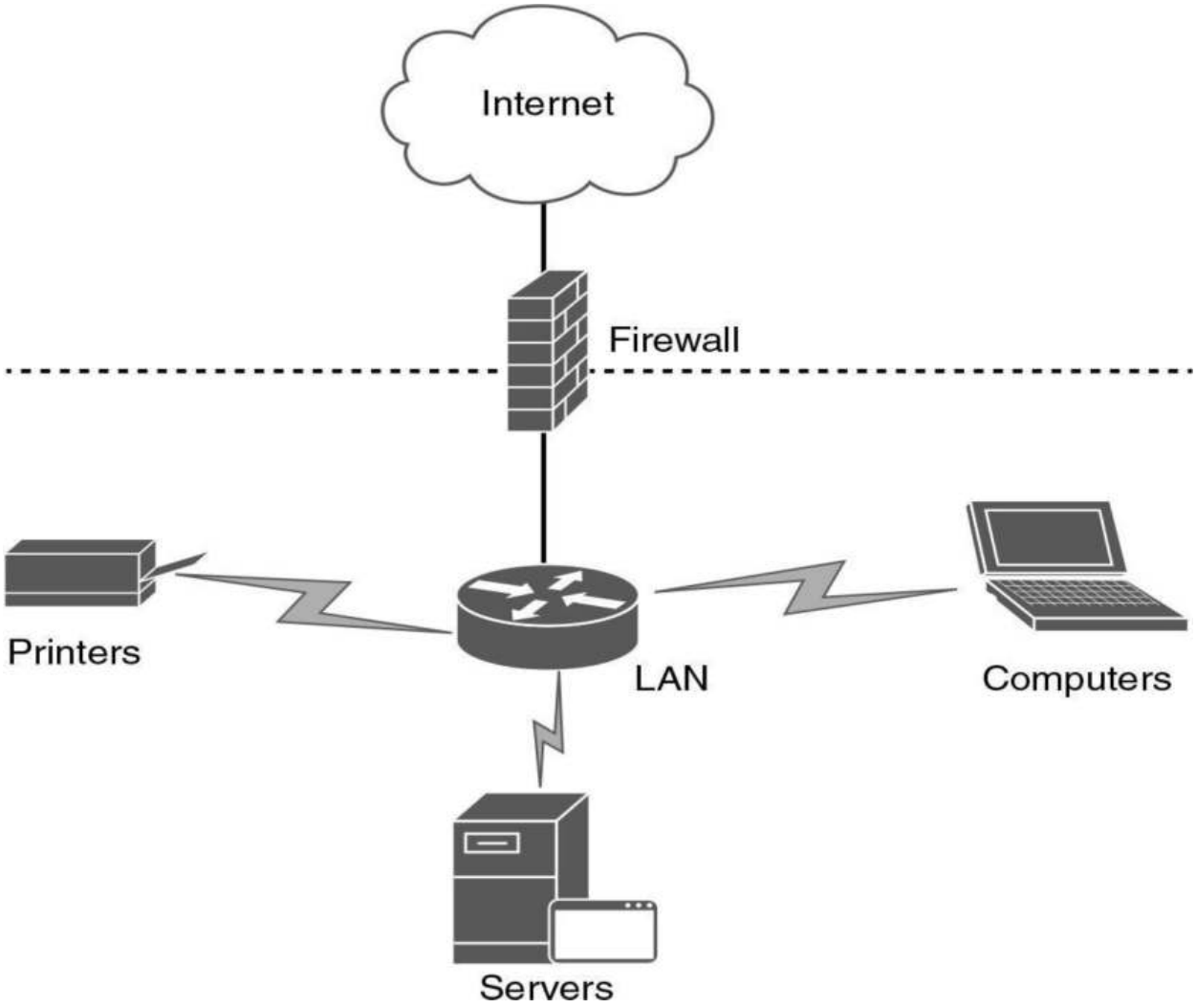


Figure 14-8 *Firewall*

The primary function of all firewalls is to screen network traffic and prevent unauthorized access between two network entities. There are several types of firewalls:

- **Packet filtering firewalls:** Individual packets are examined, although the firewall does not know the contents of a packet. These firewalls provide this security by filtering incoming packets and distinguishing between TCP/UDP traffic and port numbers. The packets are either allowed entry onto the network or denied access, based on either their source or destination address or some other static information, such as the traffic type. Figure 14-9 shows an example of a stateless firewall with a very simple rules engine that indicates which traffic is allowed and which is denied.

Rule	Source Port	Destination Port
Allow INBOUND	Any	80 (web)
Allow	Any	25 (email)
Deny	Any	22 (ssh)
Allow OUTBOUND	Any	80

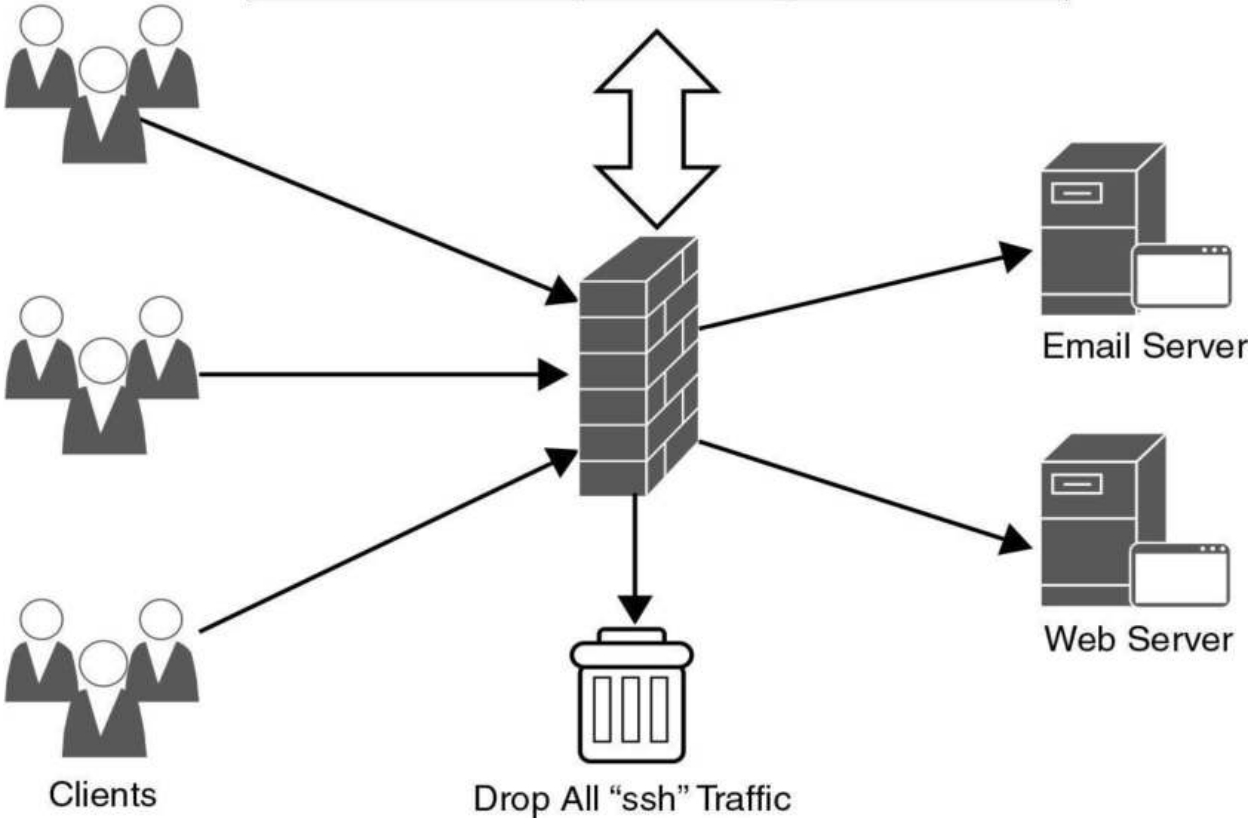


Figure 14-9 *Stateless Firewall (Packet Filtering)*

- **Stateful inspection firewalls:** Packets are examined with other packets in the flow. Such firewalls monitor the state of active connections and use this information to determine which network packets to allow. Stateful firewalls are more advanced than stateless packet filtering firewalls. They continuously track network state and active connections, such as TCP streams or User Datagram Protocol (UDP) communication. The ability to acknowledge and use the contents of incoming traffic and data packets is one of the principal advantages of stateful firewalls, as it enables these firewalls to distinguish legitimate from malicious traffic or packets. This makes stateful firewalls one of the most powerful security tools in modern policies that protect network connections by implementing additional security procedures for new or ongoing/active links. Figure 14-10 shows an example of a stateful firewall that keeps track of the data from User1 and allows it to flow to the email and web server.
- **An application-level or proxy** firewall protects network resources by filtering messages at the application layer. In addition to determining which traffic is allowed and which is denied, a proxy firewall uses stateful inspection and deep packet inspection to analyze incoming traffic for signs of attack. The key benefit of application-layer filtering is blocking specific content, such as known malware or content from individual websites. A proxy firewall can recognize when particular applications and protocols, such as Hypertext Transfer Protocol (HTTP), File Transfer Protocol (FTP), and Domain Name System (DNS), are being misused. As the firewall sees incoming packets, it inspects each protocol in the stack, noting the various states. At Layer 7 (the application layer), the firewall looks up the rules and applies them to incoming packets. Based on the system rules, it may perform other functions, such as URL filtering, data modification, logging, and object caching.

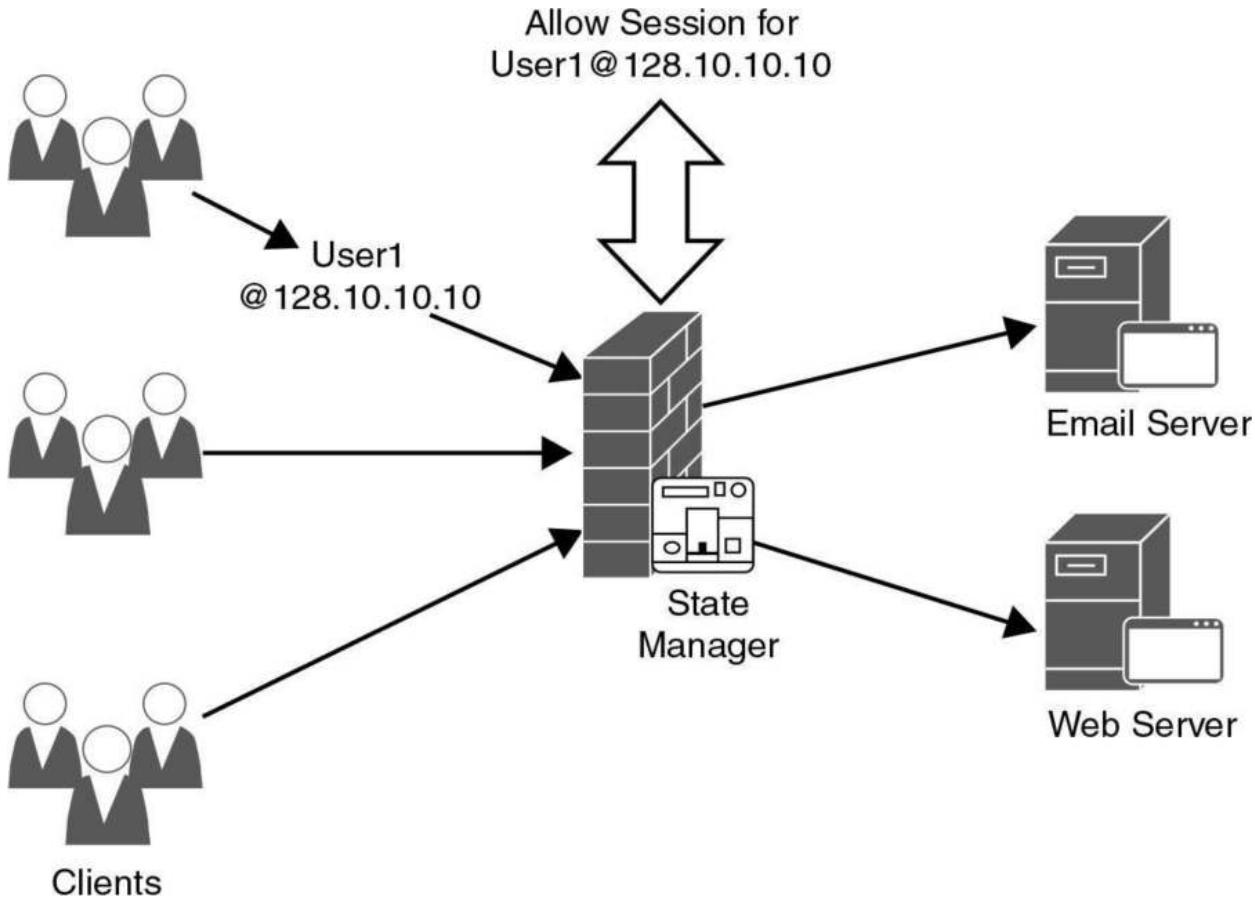


Figure 14-10 *Stateful Firewall (Context Aware)*

- **Next-generation firewall:** Some standard features of next-generation firewall architectures include deep-packet inspection (checking the actual contents of the data packet), TCP handshake checks, and surface-level packet inspection. Next-generation firewalls may also include other technologies, such as intrusion prevention, to automatically stop attacks against a network.

Intrusion Detection Systems (IDSs)

An intrusion detection system (IDS), as shown in Figure 14-11, is a passive system that monitors network traffic. In this case, the traffic is a copy of the traffic as the network sees it. In a typical network scenario, packets or flows are replicated in hardware and sent to the IDS. The IDS processes the packets and detects malicious signatures. In this case, the IDS cannot detect any suspicious activity in real-time; it must let a few packets pass before initiating any action. An IDS usually has to rely on other networking elements, such as routers or switches, to take any corrective measures to stop suspicious traffic from passing through. One of the most significant disadvantages of an IDS is that a single packet attack is almost always successful, which means it is hard to prevent these attacks.

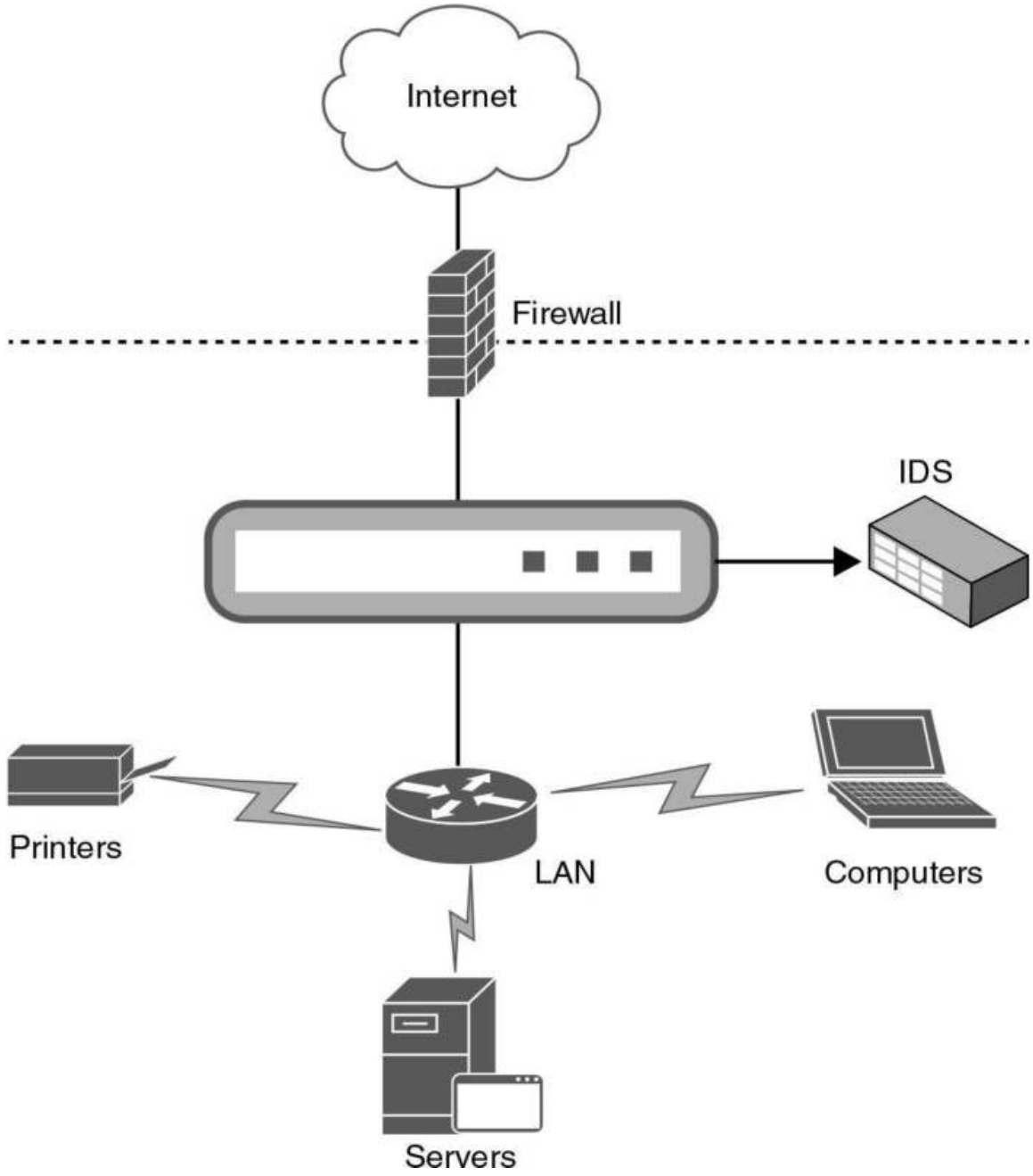


Figure 14-11 *IDS Receiving a Copy of the Packets*

Intrusion Prevention Systems (IPSs)

An intrusion prevention system (IPS), as shown in Figure 14-12, works in real-time and inline. Usually, an IPS is collocated with a network element such as a router or switch. Like an IDS, an IPS does deep-packet inspection of packets; however, an IPS takes action immediately if it determines that a packet is vulnerable. Therefore, only trusted packets are allowed into the enterprise network.

Domain Name System (DNS)

You probably don't remember many phone numbers; instead, you record them in your contacts so you can look up the name of a person you want to call. Domain Name System (DNS) works quite similarly: It keeps a mapping of domain names and the IP addresses of the servers where the domains can be reached. Each device on the Internet has a unique IP address, and DNS eliminates the need to memorize any IP addresses it uses. Figure 14-13 shows how a client request gets to the DNS resolver, which in turn gets to the primary DNS server.

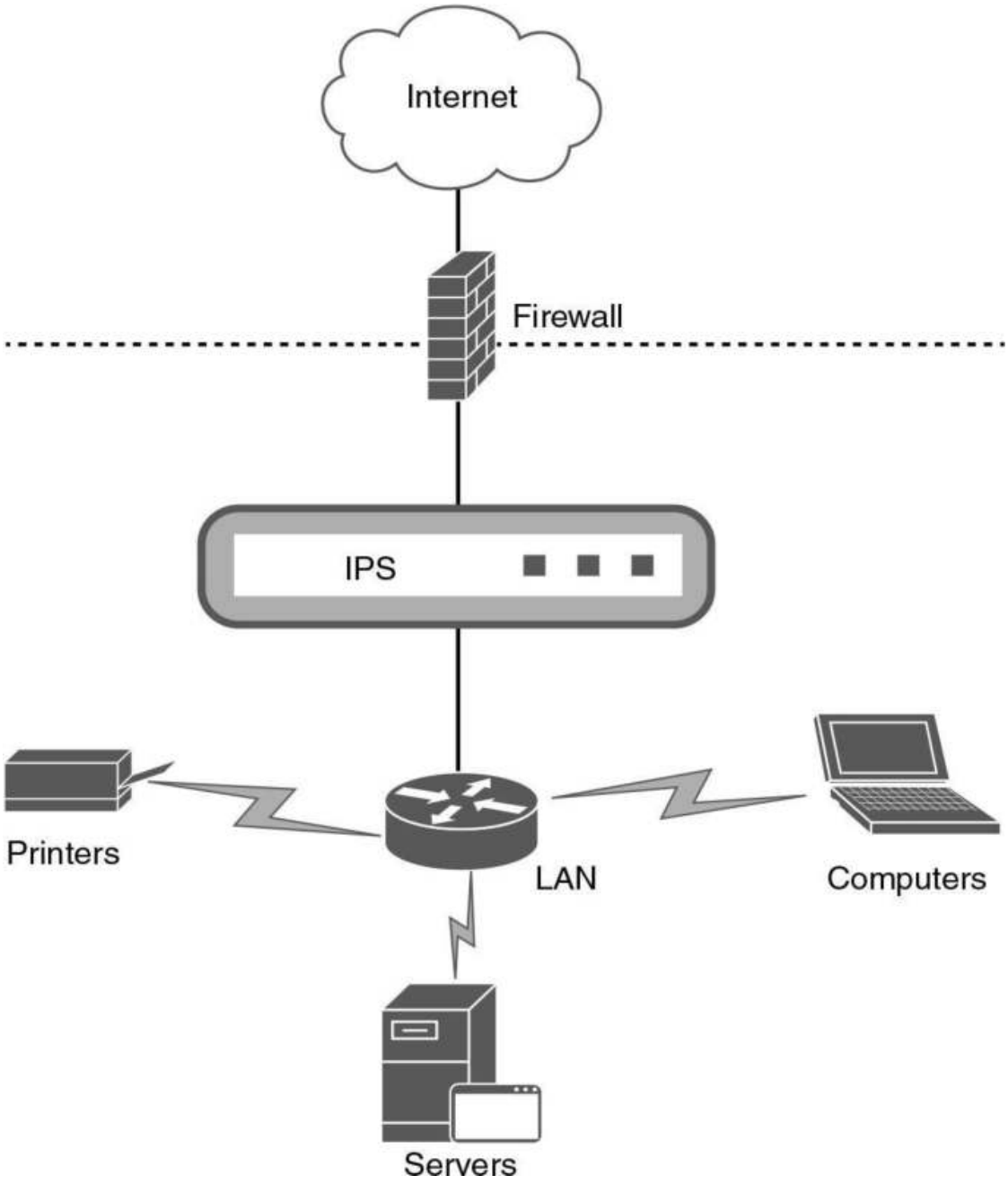


Figure 14-12 *IPS Inline with Traffic*

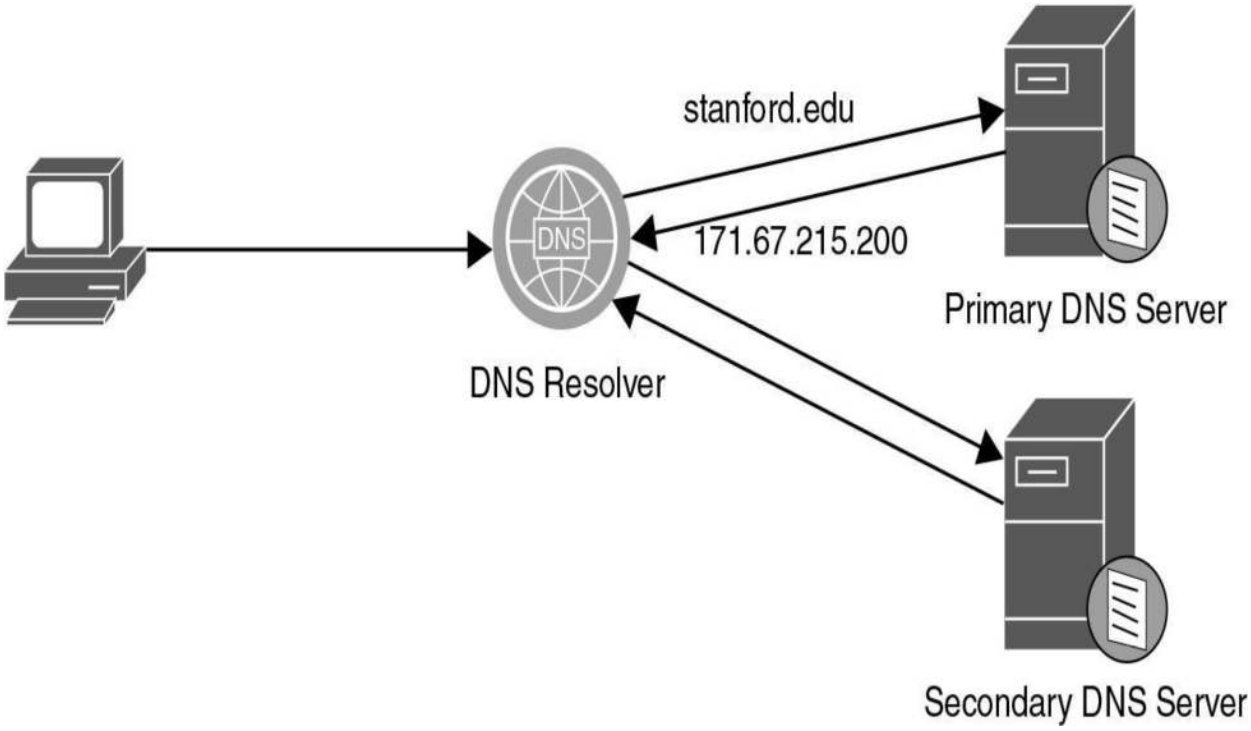


Figure 14-13 *DNS Operation*

To understand how DNS works, the following steps walk through how a client, such as a web browser, asks for a new website:

Step 1. The user clicks on a link or types a URL for a site to visit (for example, https://developer.cisco.com).

Step 2. The browser looks in the local cache to see if the name can be resolved locally on the client.

Step 3. The browser sends a query to the DNS recursive resolver to resolve the name. The DNS recursive resolver usually resides with the Internet service provider, and it can reply if it has the IP address or pass on the query to the next resolver in the chain. (There are public name resolvers like Cisco Umbrella and Google.) Eventually, one of the resolvers may respond with the IP address.

Step 4. If the DNS resolvers cannot find the IP address, the browser tries the root name servers, which forward the query to top-level domain (TLD) servers and eventually authoritative servers until results are achieved.

Step 5. When the client has the IP address for the domain, it gives the IP address to the browser.

Nslookup (Name Server Lookup) is an excellent utility for most operating systems to get details about a particular host or domain from a DNS server. The syntax for the command is as follows:

```
nslookup [-option] [name | -] [server]
```

Examples 14-4 and 14-5 show examples of using Nslookup.

Example 14-4 *Using Nslookup for a Simple Host Lookup*

nslookup stanford.edu :

nslookup with a simple domain name will return the IP address of the

stanford.edu

```
$ nslookup standford.edu
Server: 2601:647:5500:1ea:9610:3eff:fe18:22a5
Address: 2601:647:5500:1ea:9610:3eff:fe18:22a5#53

** server can't find standford.edu: NXDOMAIN

$ nslookup stanford.edu
Server: 2601:647:5500:1ea:9610:3eff:fe18:22a5
Address: 2601:647:5500:1ea:9610:3eff:fe18:22a5#53

Non-authoritative answer:
Name: stanford.edu
Address: 171.67.215.200
```

Example 14-5 *Using Nslookup to Get Information About a Domain*

```
nslookup -type=any stanford.edu :

The -type=any parameter allows us to get all the DNS records for that domain,

including the mail servers, etc.


$ nslookup -type=any stanford.edu

;; Truncated, retrying in TCP mode.

Server: 10.168.243.90

Address: 10.168.243.90#53


Non-authoritative answer:

stanford.edu mail exchanger = 10 mxa-00000d03.gslb.pphosted.com.

stanford.edu mail exchanger = 10 mxb-00000d03.gslb.pphosted.com.

stanford.edu nameserver = ns5.dnsmadeeasy.com.

stanford.edu nameserver = ns6.dnsmadeeasy.com.

stanford.edu nameserver = ns7.dnsmadeeasy.com.

stanford.edu nameserver = argus.stanford.edu.

stanford.edu nameserver = atalante.stanford.edu.

stanford.edu nameserver = avallone.stanford.edu.

stanford.edu

  origin = argus.stanford.edu

  mail addr = hostmaster.stanford.edu

  serial = 2019138199

  refresh = 1200

  retry = 600

  expire = 1296000

  minimum = 1800
```

Load Balancing

Load balancing is the process of distributing user requests across multiple servers. The goal is to ensure that no single server bears too much demand. In essence, load balancers are like virtual servers, receiving all user requests and forwarding these requests based on the load balancer's policy to one of the servers that host the website. The load balancer attempts to ensure that each server receives a similar number of requests. Many load balancers can monitor servers and compensate for any servers that become unavailable. By spreading the work evenly, load balancing improves application responsiveness.

Figure 14-14 shows how requests from various clients can be load-balanced (or distributed) to various servers.

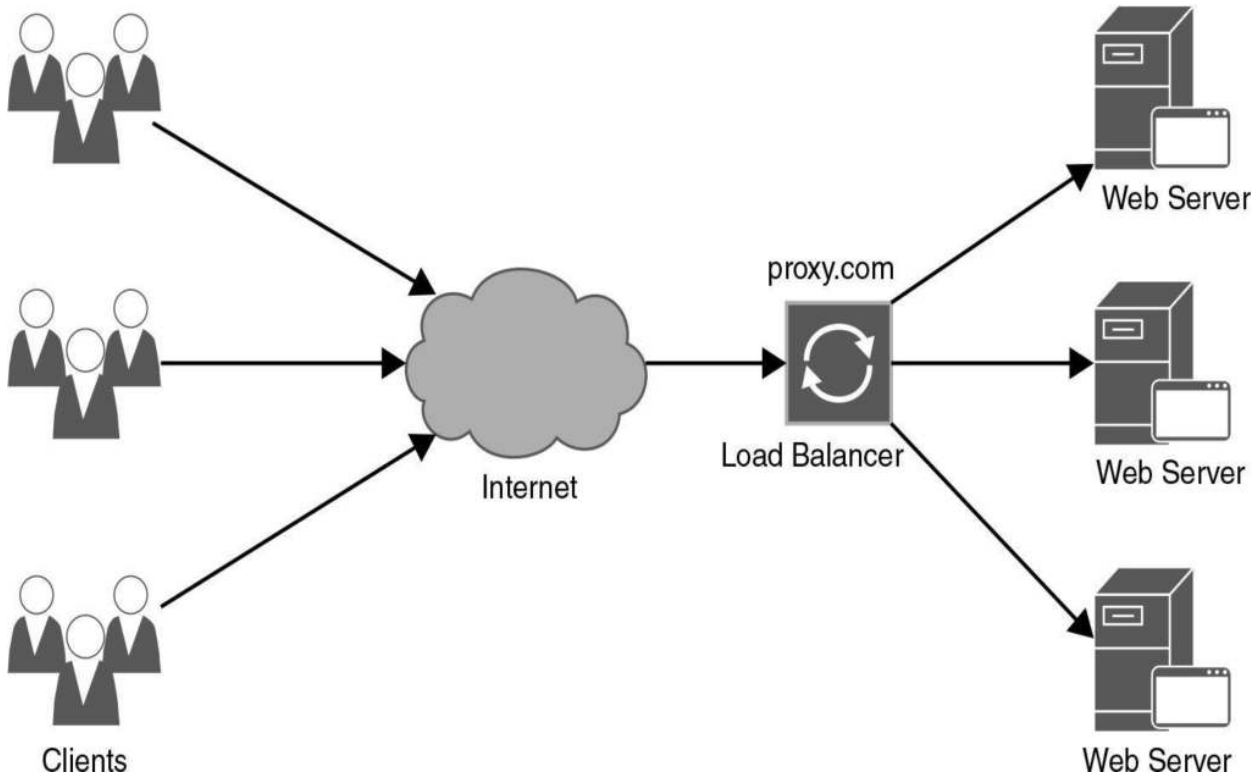


Figure 14-14 *Load Balancing*

Load balancing uses algorithms such as the following:

- **Round-robin:** Selects servers in turn
- **Least connected:** Select the server with the lowest number of connections; this is recommended for more extended sessions
- **Source/IP-hash:** Chooses a server based on a hash of the source IP
- **Cookie marking:** Adds a field in the HTTP cookies, which could be used for decision-making
- **Consistent IP-hash:** Adds and removes servers without alarming cached items or session persistence

A reverse proxy accepts a request from a user, forwards it to a server that can fulfill it, and returns the server's response to the client. A reverse proxy can include some or all of the following functionality:

- **Security:** The web or application servers are not visible from the external network, so malicious clients cannot access them directly to exploit vulnerabilities. Many reverse proxy servers include features that help protect backend servers from distributed denial-of-service (DDoS) attacks—, for example, by rejecting traffic from particular client IP addresses (blacklisting) or limiting the number of connections accepted from each client.
- **Scalability and flexibility:** Clients see only the reverse proxy's IP address. This is particularly useful in a load-balanced environment, where you can scale the number of servers up and down to match fluctuations in traffic volume.
- **Web acceleration:** Acceleration, in this case, means reducing the time it takes to generate a response and return it to the client. Some of the techniques for web acceleration include the following:
- **Compression:** Compressing server responses before returning them to the client (for instance, with gzip) reduces the required bandwidth, which speeds their transit over the network.
- **SSL termination:** Encrypting the traffic between clients and servers protects it as it crosses a public network such as the Internet. However, decryption and encryption can be computationally expensive. By decrypting incoming requests and encrypting server responses, the reverse proxy frees up resources on backend servers, which the servers can then devote to their primary purpose—serving content.
- **Caching:** Before returning the backend server's response to the client, the reverse proxy stores a copy of it locally. When the client (or any other client) makes the same request, the reverse proxy can provide the response from the cache instead of forwarding the request to the backend server. This decreases the client's response time and reduces the backend server's load. This works great for "static" content, but new techniques can also be used for "dynamic" content.
- **Content filtering:** This involves monitoring traffic to and from the web server for potentially sensitive or inappropriate data and taking action as necessary.

- **Authentication:** The reverse proxy authenticates users via various mechanisms and controls access to URLs hosted on the web server.

Figure 14-15 shows how requests come in from various clients. The reverse proxy can terminate an SSL connection or even inspect incoming traffic.

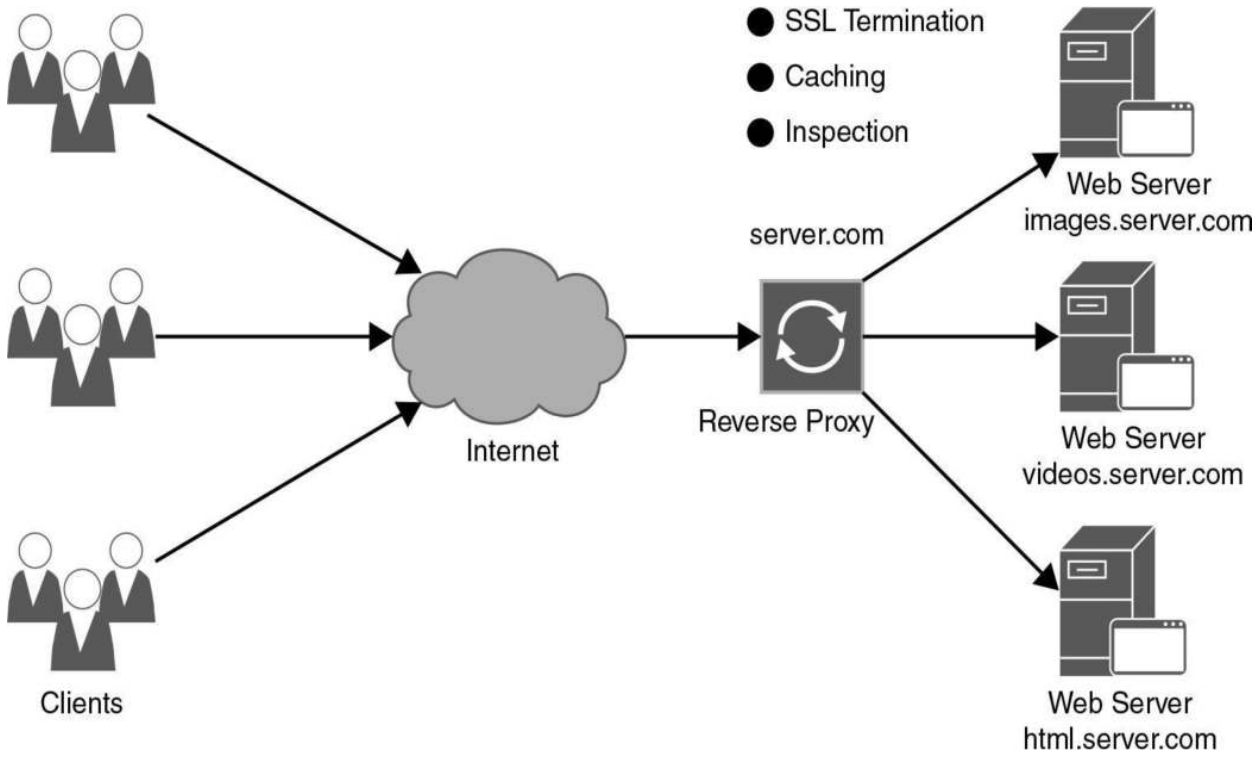


Figure 14-15 *Reverse Proxy*

Figure 14-16 illustrates the reverse proxy concept with a user requesting a web page from https://server.com. In this case, server.com is the reverse proxy, which first terminates SSL, then translates the request and issues a new request to the images.server.com server. The image server then responds to the request, gathering and translating the response and sending it back to the client. The translation could mean applying compression (gzip) to the response.

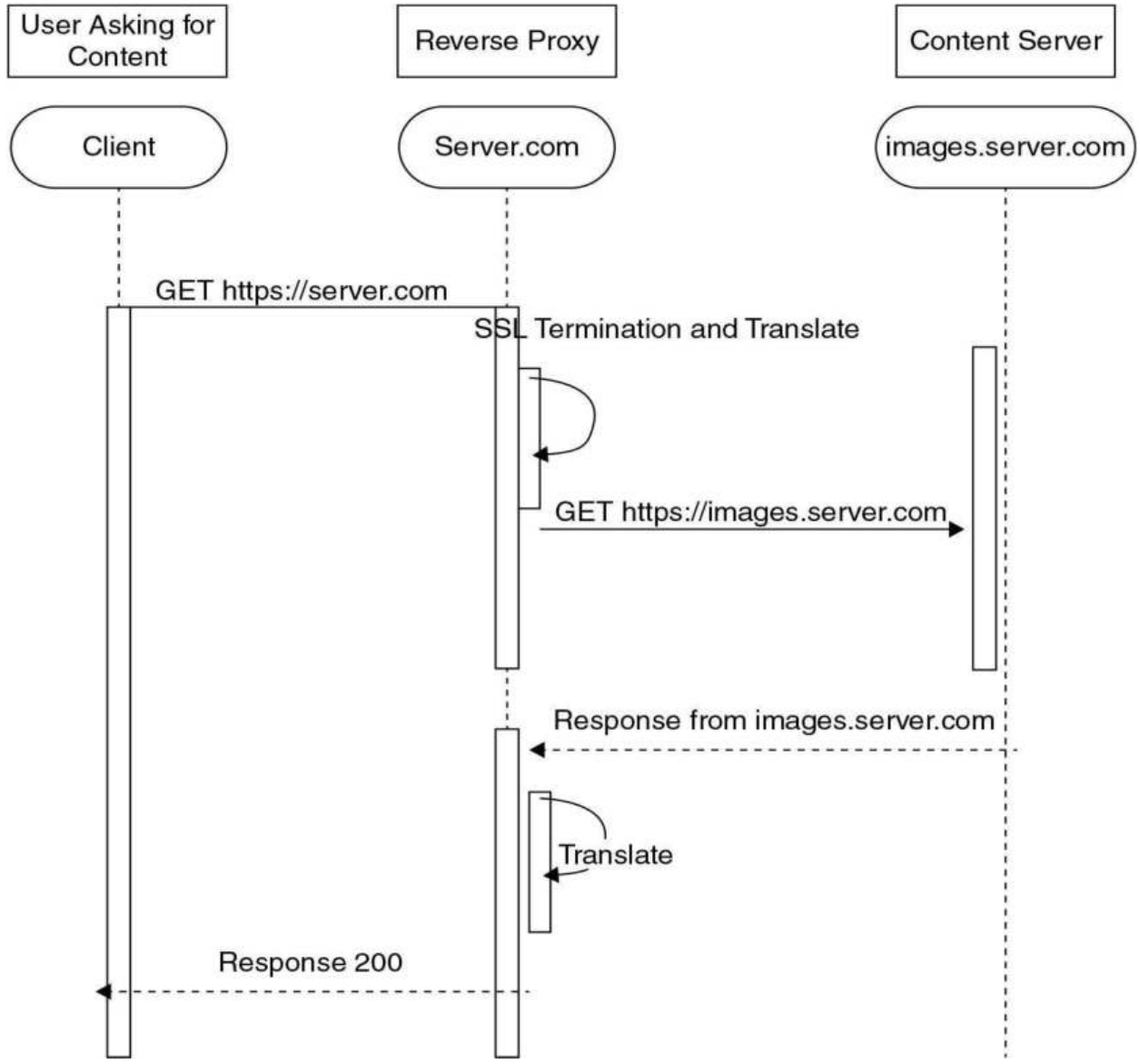


Figure 14-16 *Reverse Proxy Flow Diagram*

Exam Preparation Tasks

As mentioned in the section "How to Use This Book" in the Introduction, you have a couple of choices for exam preparation: the exercises here, Chapter 19, "Final Preparation," and the exam simulation questions on the companion website.

Review All Key Topics

Review the most important topics in this chapter, noted with the Key Topic icon in the outer margin of the page. Table 14-4 lists these key topics and the page number on which each is found.

Table 14-4 *Key Topics for Chapter 14*

Key Topic Element	Description	Page
List	Assets, threats, vulnerabilities, and risks	
Table 14-2	Threats and mitigation	
Paragraph	Open Web Application Security Project (OWASP)	
Paragraph	Nmap	
Figure 14-6	Data security	
Paragraph	Firewalls	
Paragraph	Intrusion detection system	
Paragraph	Domain Name System	
List	Reverse proxy	

Define Key Terms

Define the following key terms from this chapter and check your answers in the glossary:

Secure Sockets Layer (SSL)
Transport Layer Security (TLS)
Open Web Application Security Project (OWASP)
OWASP Top 10
Nmap
multifactor authentication (MFA)
cryptography
Rivest–Shamir–Adleman (RSA)
software development lifecycle (SDLC)
Dlint
Domain Name System (DNS)
intrusion detection system (IDS)
intrusion prevention system (IPS),

Chapter 15. Infrastructure Automation

This chapter covers the following topics:

- **Controller Versus Device-Level Management:** This section compares and contrasts the two network device management options currently available: controller-based and device-level management.
- **Infrastructure as Code:** This section introduces the concepts of infrastructure as code.
- **Continuous Integration/Continuous Delivery Pipelines:** This section examines CI/CD pipelines, why they are used, the problems they address, and how they can be applied to infrastructure automation.
- **Automation Tools:** This section covers popular automation tools that are used for configuration management and network automation.
- **Cisco Network Services Orchestrator (NSO):** This section provides an introduction to Cisco NSO.
- **Cisco Modeling Labs (CML):** This section explains what Cisco CML is and how it can be used to build network simulations.
- **Python Automated Test System (pyATS):** This section examines pyATS, including the components that make up the testing solution and how it can be used to automate network testing.

This chapter covers infrastructure automation concepts. It starts with a comparison of management of infrastructure on a device-by-device basis or using a central controller. Thanks to virtual devices and public cloud technologies, infrastructure can be created, modified, and decommissioned dynamically by using the concept of infrastructure as code (IaC). This chapter introduces infrastructure as code and the problems it solves. This chapter also covers continuous integration/continuous delivery (CI/CD) pipelines and their importance in the software development lifecycle. In addition, the chapter covers automation tools and introduces the popular configuration management tools Ansible and Terraform. This chapter then discusses Ansible concepts such as control nodes, playbooks, and modules; and Terraform concepts such as providers, resources, data sources, state, and modules. Next, the chapter covers Cisco NSO, including the architecture of the platform, common use cases, the network simulation feature, and the use of tools such as **curl** and Postman. Finally, the chapter provides overviews of Cisco CML and pyATS.

"Do I Know This Already?" Quiz

The "Do I Know This Already?" quiz allows you to assess whether you should read this entire chapter thoroughly or jump to the "Exam Preparation Tasks" section. If you are in doubt about your answers to these questions or your own assessment of your knowledge of the topics, read the entire chapter. Table 15-1 lists the major headings in this chapter and their corresponding "Do I Know This Already?" quiz questions. You can find the answers in Appendix A, "Answers to the 'Do I Know This Already?' Quiz Questions."

Table 15-1 *"Do I Know This Already?" Section-to-Question Mapping*

Foundation Topics Section	Questions
Controller Versus Device-Level Management	1, 2
Infrastructure as Code	3, 4
Continuous Integration/Continuous Delivery Pipelines	5, 6
Automation Tools	7–9
Cisco Network Services Orchestrator (NSO)	10
Cisco Modeling Labs (CML)	11
Python Automated Test System (pyATS)	12

Caution

The goal of self-assessment is to gauge your mastery of the topics in this chapter. If you do not know the answer to a question or are only partially sure of the answer, you should mark that question as wrong for purposes of self-assessment. Giving yourself credit for an answer that you correctly guess skews your self-assessment results and might provide you with a false sense of security.

1. What protocols were historically used to manage network devices? (Choose two.)
   - **a.** SMTP
   - **b.** SSH
   - **c.** TFTP
   - **d.** Telnet
   - **e.** SNMP
2. Which of the following is an example of a central network controller?
   - **a.** Nagios
   - **b.** Grafana
   - **c.** Cisco Catalyst Center (formerly Cisco DNA Center)
   - **d.** Prometheus
3. What are two common approaches to infrastructure as code?
   - **a.** Subjective
   - **b.** Declarative
   - **c.** Imperative
   - **d.** Objective
4. Which of the following are issues with deploying infrastructure that are being addressed with infrastructure as code? (Choose two.)
   - **a.** The amount of time needed to deploy new infrastructure
   - **b.** Maintaining code across different versions
   - **c.** Troubleshooting issues
   - **d.** Lack of code documentation
5. What is a key component of a continuous integration pipeline?
   - **a.** Integration server
   - **b.** Delivery server
   - **c.** File server

d. Build server

6. What are the four steps in a CI/CD pipeline?

   a. Code, test, build, deploy

   b. Source, build, test, deploy

   c. Source, test, deploy, monetize

   d. Build, test, deploy, monetize

7. What command is used to run Ansible playbooks?

   a. ansible

   b. ansible-playbook

   c. ansible-playbooks

   d. ansible-plays

8. What syntax is typically used for Terraform configuration files?

   a. Python

   b. HCL (HashiCorp Configuration Language)

   c. YAML

   d. Java

9. What Terraform command previews proposed infrastructure changes?

   a. terraform test

   b. terraform plan

   c. terraform apply

   d. terraform dry-run

10. What Cisco NSO component manages the YANG data models that get sent to a device?

    a. Service Manager

    b. Device Manager

    c. Core Engine

    d. CDB

11. What language are Cisco CML topology files written in?

    a. JSON

    b. CSV

    c. YAML

    d. XML

12. What are the two main components of pyATS?

    a. pyATS test framework

    b. pyATS repo

    c. pyATS testbed

    d. pyATS library

Foundation Topics

Controller Versus Device-Level Management

Historically, network devices were managed through command-line interfaces (CLIs) by using protocols such as Telnet and Secure Shell (SSH). The network administrator would connect to the CLI of a network device using a software client that implements these protocols and perform the configuration changes needed for the device to function as expected. On a small- to medium-sized network, this approach might still work, but as networks keep growing bigger and more and more devices become connected, managing networks in a device-by-device manner becomes time-consuming and prone to errors. Combine this with the needs for automation, network programmability, and reductions in operational costs, and you can see that a new way of managing networks is needed.

Network controllers are a possible solution to the challenges of managing networks in a device-by-device manner. A network controller is a centralized software platform dedicated to managing the configuration and operational data of network devices. Controllers take over the management of network devices, meaning that all interactions with the devices have to go through the controller. They provide an abstraction layer between the administrators of the network and network devices. Network controllers usually expose a northbound REST API interface for management and integration with third-party systems and one or several southbound interfaces through which they connect and manage the network devices. Typical southbound interfaces are CLI-based interfaces that use Telnet and SSH, OpenFlow, SNMP, NETCONF, RESTCONF, and so on. In most situations, the network controller manages different network devices running different network operating systems with different versions and different features; by abstracting the underlying network capabilities, the controller exposes only a subset of the network functions. This minimal loss of network functionality is an assumed risk when choosing to go the network controller route for managing a network.

On one side is the controller-based management of networks supporting a subset of network functions, and on the other side is direct access to the devices with full access to all network features. Network administrators have to choose between these two options, keeping in mind the requirements of the network and which option works best for each situation.

There are several other considerations to keep in mind when integrating controllers into a network. Support for atomic network configuration is one of them. Atomicity in the case of network configuration means that either the intended configuration has been applied to all network elements without error or; to deal with potential errors, a rollback mechanism is in place to ensure that the partial configuration is removed and the devices are brought back to the initial state before any attempt is made to change the configuration. Atomicity is very important to ensure a holistic and uniform view of the network and to make sure that partial and incomplete configurations do not negatively affect the functionality of the network.

Another consideration with network controllers involves auditing and managing the configuration drift of the network elements. Configuration drift happens when the configuration of a network device has changed from its intended state, which means the configuration of the device in the network controller is different from the configuration of the device itself. This situation can easily lead to inconsistencies and unexpected network behavior. A good controller has built-in mechanisms to address configuration drift; in some instances, these mechanisms go as far as limiting or even blocking device configuration changes.

The main advantage provided by network controllers is that the abstraction and central administration of individual network devices eliminate the need for device-by-device configuration, management, and monitoring, as shown in Figure 15-1.

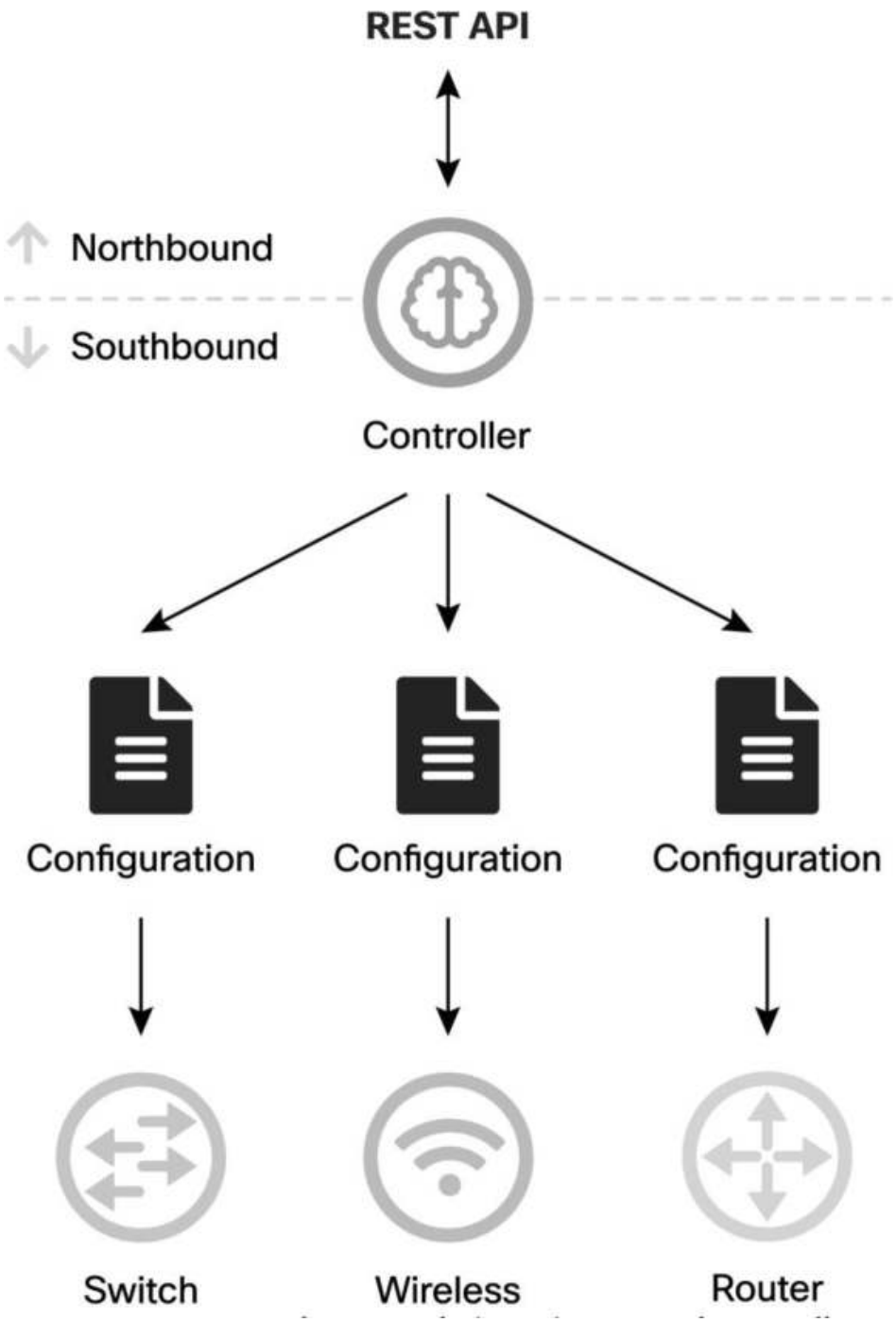


Figure 15-1 *Network Managed Through a Network Controller*

Key Topic

Direct device configuration allows access to the full feature set of a device, but it can be time-consuming and prone to errors—especially in large networks. The scale of a network is certainly a limiting factor when performing configuration and management tasks on a device-by-device basis. As you've seen in Chapter 12, "Model-Driven Programmability," historically, there has been an evolution from custom CLIs to standard data models and network configuration protocols such as NETCONF and RESTCONF. Having standardized protocols for network configuration as well as standard data models definitely helps with the network automation efforts and the scale limitations. Configuration changes made on a single device can be replicated across an entire network. Device-by-device configuration is illustrated in Figure 15-2.

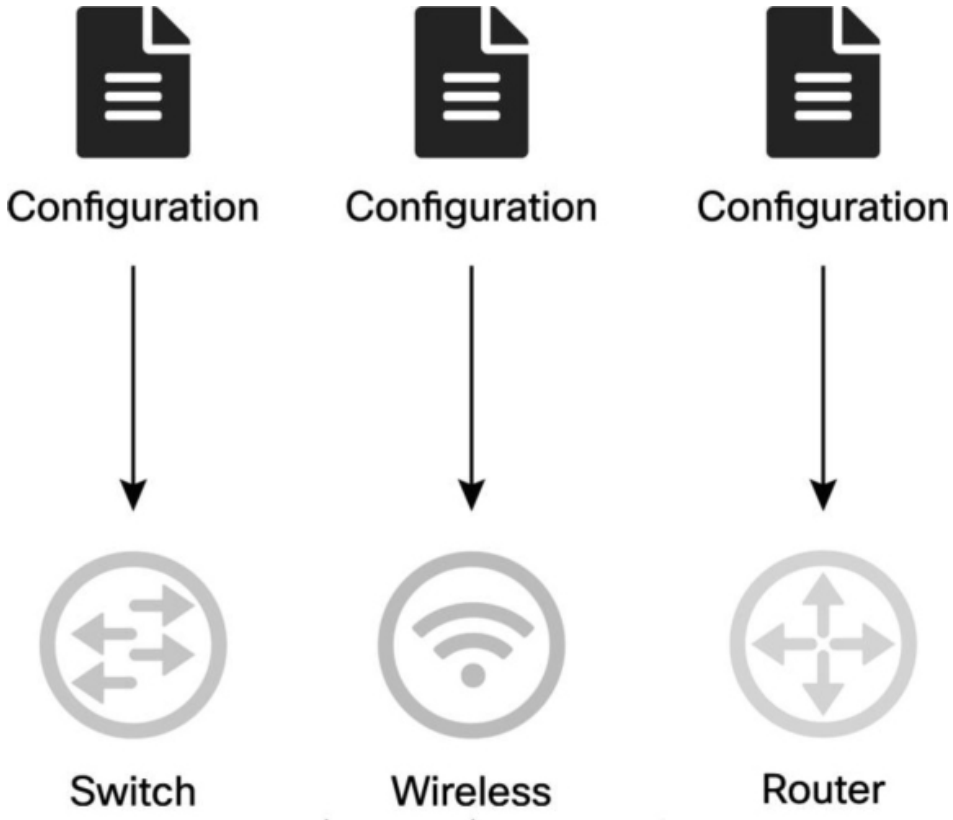


Figure 15-2 *Network Managed on a Device-by-Device Basis*

The use of a controller for network management and device-by-device configuration are not mutually exclusive. A controller can be used to globally configure devices, while direct connection to devices can be helpful for monitoring and ensuring that the changes made by the controller do not create undesired behavior.

Chapter 8, "Cisco Enterprise Networking Management Platforms and APIs," provides an example of a network controller with Cisco Catalyst Center (formerly Cisco DNA Center). Cisco Catalyst Center can be used to completely configure, manage, and monitor networks. It can also be used to deploy SD-Access fabrics. It is possible to deploy SD-Access by using automation tools and connecting to each device using NETCONF, the CLI, or even SNMP. This capability gives the network administrator more granular control but at the same time requires more work and time to define configuration parameters and data models and to ensure that the correct configuration gets applied to all the devices in a unified manner. Cisco Catalyst Center performs all these tasks automatically, and automation scripts can be used to collect operational data and ensure that the network performs within optimal parameters.

Infrastructure as Code

Inspired by software development practices, infrastructure as code is a new approach to infrastructure automation that focuses on consistent, repeatable steps for provisioning, configuring, and managing infrastructure. For a long time, infrastructure has been provisioned and configured in a manual fashion. Deploying a new application on a network used to take anywhere from weeks to months to even years in some situations. During that time, the IT team would try to size the hardware requirements for the new application, go through the approval process to purchase the new hardware, order the hardware, wait for it to be delivered, rack it up, and start the provisioning and configuration steps. The provisioning and configuration of the new hardware was, in most cases, a manual process: installing the operating system one step at a time following guided instructions and then finally installing and configuring the new application. At best, technologies like PXE booting and installation scripts would be used to try to automate the arduous process of provisioning new hardware and installing new applications. This approach resulted in one-of-a-kind, or "snowflake," environments in which each application ran on different hardware, with different options and features enabled; supporting and maintaining such environments was incredibly challenging. A new paradigm for deploying and managing infrastructure was needed.

The IT infrastructure of the past had a number of limitations, including high deployment and maintenance costs, long wait times for new infrastructure to be ready for production (slowing down new application deployments and hampering innovation and fast prototyping), and difficulty in troubleshooting because each environment was slightly different. Several technologies have evolved to address these limitations. The one that has had the biggest impact is public cloud computing. Besides making infrastructure available almost instantaneously and reducing the costs associated with purchasing hardware, cloud computing has also changed the way IT infrastructure is provisioned, configured, and consumed. With the advent of APIs, public cloud infrastructure can more easily be automated, and cookie-cutter templates can be created for easy and fast deployment of new infrastructure.

Infrastructure as code is all about the representation of infrastructure through machine-readable files that can be reproduced for an unlimited amount of time. It is a common practice to store these files in a version control system like Git and then use them to instantiate new infrastructure—whether servers, virtual machines, or network devices. In this way, infrastructure can be treated as source code, with versioning, history, and easy rollback to previous versions, if needed. Being able to spin up new infrastructure in a repeatable, consistent fashion becomes extremely useful, for example, in cases in which there is a spike in traffic and in the number of application requests for a certain period of time. In such cases, new servers, load balancers, firewalls, switches, and so on can be dynamically provisioned within seconds to address the additional load. Once the traffic subsides and the need for higher capacity subsides, the infrastructure can be scaled down by dynamically decommissioning servers, load balancers, and so on to save on costs. This *elasticity* of the infrastructure is another of the benefits of defining all infrastructure as code.

When the infrastructure is defined through code, the infrastructure becomes transparent in the sense that it is enough to read the code in the document to know all the characteristics and features that will be instantiated when it gets provisioned. Other people can review the code, make improvements, have the changes tracked through version control, and make sure the code is in compliance with the security requirements of the company.

While change was frowned upon in the infrastructure of the past, and a limited number of strict change windows were provided through the year, with infrastructure as code, change is welcomed and even desired. With infrastructure as code, change is a catalyst to improve the reliability and the performance of the infrastructure as a whole. Just as source code goes through multiple versions and becomes better with each new release, infrastructure becomes more resilient and reliable with each new version.

Through the use of infrastructure as code, it is possible to have identical environments for testing, integration, and production. This capability leads to significantly fewer troubleshooting steps needed to repair and maintain the infrastructure and the applications running on top of it.

There are usually two types of approaches to infrastructure as code:

- **Declarative:** With the declarative approach, the desired state of the system is defined and then the system executes all the steps that need to happen to attain the desired state.
- **Imperative:** The imperative approach defines a set of commands that have to be executed in a certain order for the system to achieve the desired state.

A popular infrastructure as code solution is Terraform from HashiCorp. Infrastructure as code integrates very well in the continuous integration/continuous delivery pipelines discussed next.

Continuous Integration/Continuous Delivery Pipelines

Software is quickly becoming pervasive in all aspects of our lives. From smartphones to smart cars and smart homes, we interact with software hundreds or even thousands of times each day. Under the DevOps umbrella, several efforts have been made to improve software development processes to increase the speed, reliability, and accuracy of software development. *Continuous integration/continuous delivery (CI/CD) pipelines* address all these requirements and more. All the software development processes—from writing code to building, testing, and deploying—were manually performed for years, but CI/CD pipelines can be used to automate these steps. Based on specific requirements for each company, different tools and different solutions are used to implement these pipelines. Some companies implement only continuous integration solutions, and others take advantage of the whole CI/CD pipeline, automating their entire software development process.

CI/CD is a series of automated steps that code goes through from the IDE of the individual developer, through building, testing, and finally deployment to staging and production environments. Although usually used together, continuous integration and continuous delivery are two separate concepts that can be addressed separately.

Continuous integration is a software development practice in which developers commit their code to a central repository that is part of a version control system on an hourly, daily, or weekly basis or at some other frequency that is agreed upon within the team. As developers commit their code to the central repository, it is important to verify that the code integrates well with the full code base.

As new code is written, it is critically important to run code base tests against the new changes in the code base in a fast, automated fashion. Continuous integration tools accomplish this task exactly. As the code gets committed to a central repository —in most cases, to a Git-style version control system—a webhook gets triggered; it starts the whole automated testing phase of

the pipeline. Common triggers at this stage include automatic integrations with tools such as Webex for immediate notifications on the status of the tests. In the event that a test fails, the developer is notified and can correct the issues that caused the test to fail very early in the development process. If all the tests pass, the code moves on to the next stage of the pipeline.

A key component of any CI pipeline is the build server. The role of the build server is to react to developers committing their code to the central repository and to start the initial tests on the new code features. Most version control systems support webhook mechanisms to automatically notify the build server when pull requests are opened and code is committed. Popular build servers include Jenkins, Travis CI, and Drone CI. Table 15-2 compares development environments that have implemented CI pipelines and the ones that have not.

Table 15-2 *Development Environments With and Without CI*

Development Without CI	Development With CI
Increased number of bugs	Fewer bugs
Insufficient testing	Automated builds, tests, documentation, and reports
Few releases per year	Multiple releases per day
Lack of integration testing	Dedicated build and test servers
Project delays	Frequent commits
Fewer features	More features
Instability	Stability

Continuous delivery adds on top of continuous integration all the remaining steps needed to automate the entire software release cycle, from building through testing to a deployable release (see Figure 15-3).

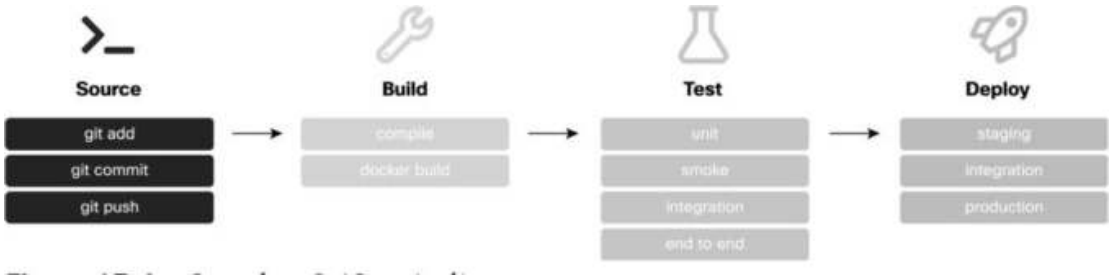


Figure 15-3 *Complete CI/CD Pipeline*

In the building phase, the code is compiled for languages such as Java, C/C++, or Go; for code written in interpreted languages such as Ruby or Python, the compilation step can be ignored. Regardless of the language the code is written in, often, the output of the building phase is a container image that contains all the code, all required dependencies, and startup scripts to ensure that the application runs consistently, whatever the destination environment might be.

During the testing phase, automated tests are run to ensure correct behavior of the product. In an effort to catch software bugs as soon as possible, tests are developed at this stage to make sure the new code follows the expected requirements. Several types of tests are run at this stage, from unit and smoke tests that perform quick sanity checks, to integration, code coverage, code standards, and end-to-end tests that try to mimic the way users interact with the software as closely as possible. Developers follow the results of the tests, and if any of the tests fail, the developers can immediately address the problems and fix the code and restart the pipeline until all tests pass.

The next phase in the CI/CD pipeline is the deployment phase. By this stage, the software has been packaged and tested and is ready to be deployed for consumption. Before containers, in most cases, this meant installing the new binaries in new servers, either physical or virtual, and reconfiguring the load balancers to point to the new servers and monitor the environment to ensure

that the new version worked as expected. With the advent of Docker and similar container technologies, the deployment phase has become much easier to implement. Containers can be destroyed and restarted with the new version of code within seconds. Container orchestrator solutions, such as Kubernetes, can be used to simplify this process even further. By using the container orchestrator APIs, developers can make a call to roll out the new Docker containers with the new features—while deployment strategies can minimize or avoid downtime. Several strategies can be used to minimize deployment downtime while deploying containers; changing only one container at a time from a pool of many instances is one of them.

In most situations, several environments are available at the deployment stage of the pipeline, and commonly include staging and production. In the staging environment, final preparations and manual testing by the product team take place to ensure that everything functions as expected. In continuous delivery, approved changes in the staging environment are ready for production deployment after a manual decision; continuous deployment automates that release.

The CI/CD pipelines described so far can be adapted for network management and configuration. Taking advantage of infrastructure as code, network configurations can be stored in version control systems, and build servers can be used to integrate with solutions such as Cisco CML to dynamically instantiate virtual networks, test the configuration changes in the virtual network, and, if the tests pass, use automation solutions such as Ansible to perform the desired changes in the production network. A working network management CI/CD pipeline could include the following:

- GitHub.com as a cloud-based version control system or Gogs as a self-hosted Git option could be used to store the configurations.
- Travis CI as a cloud-based build server or Drone as a self-hosted build server could be used to orchestrate the whole CI/CD pipeline.
- Cisco CML could be used to create a test network and verify the impact of the configuration changes in the network.
- Ansible could be used to automate the configuration of all the elements in the network.

CI/CD pipelines are used by more and more companies to speed up and increase the reliability and accuracy of their software development processes. Companies that have adopted CI/CD pipelines as part of their development processes release hundreds of software features every week.

Automation Tools

It is common practice for network administrators to perform configuration, monitoring, and maintenance activities every day on all the devices in a network or at least a subset of them. These changes were traditionally deployed manually: The network administrator would connect to each network device individually and perform the changes. The changes could be anything, from adding a new access VLAN throughout the network, to updating entries in an access control list, to changing SNMP communities' names. Where a manual task is being performed, there is an opportunity to improve the process by automating that task. Automation tools such as Ansible and Terraform can dramatically reduce the number of manual interactions with a network. These tools enable automation at scale for application deployment, infrastructure management, and network configurations. The following sections discuss each of them in turn.

Ansible

Ansible is a configuration management and orchestration tool that can be used for a variety of purposes. It can be used to configure and monitor servers and network devices, install software, and perform more advanced tasks such as continuous deployments and zero-downtime upgrades. It was created in 2012 and acquired by Red Hat in 2015. Ansible is appropriate for both small and large environments and can be used for managing a handful of servers and network devices or for managing thousands of devices. It is agentless, meaning no software or service needs to be installed on the managed device. Ansible connects to managed devices just as a regular system or network administrator would connect for management purposes—in most cases over SSH, but NETCONF and REST API interfaces are also supported. Ansible is open source and was developed using Python. The current commercial offering is Red Hat Ansible Automation Platform, whose automation controller includes a web user interface, a northbound REST API, role-based access control, statistics, and much more.

Several Ansible concepts need to be discussed before we go further:

- **Control node:** The control node is any machine that has Ansible installed. Most UNIX-like systems with Python can serve as control nodes. While Microsoft Windows is not directly supported as a control node server, Windows Subsystem for Linux (WSL) or a container can be used for development; the Ansible documentation does not support WSL for production use. Multiple control nodes can run at the same time in the same environment. It is a best practice to place the control nodes close to the systems that are being managed by Ansible to avoid network delays.

- **Managed node:** The managed nodes in the Ansible taxonomy are the network devices or servers that are managed by Ansible. They are also called hosts and do not need to have Ansible installed on them or even Python, as discussed later in this chapter.
- **Task:** Ansible tasks are the units of action. You can run them as ad hoc commands by invoking them as follows:

```
$ ansible [pattern] -m [module] -a "[module options]"
```

- **Playbook:** An Ansible playbook is a file that contains an ordered set of tasks that will be run in the order in which they are defined and as many times as needed. Playbooks are written in YAML, which makes them easy to read, write, and share. It is common (but not mandatory) to name the main playbook site.yml.
- **Inventory file:** The inventory file is a list of all the managed nodes. Also called the hostfile, it contains a list of IP addresses or hostnames for all the managed nodes as well as credentials and variables that can be referenced in playbooks. Managed devices can be grouped together by function or location or based on some other feature. As the hostfile grows larger in a bigger environment, it is a best practice to move the variables to dedicated files in group_vars/ and host_vars/ folders. The group_vars folder would contain files with definitions for variables related to groups of devices, and the host_vars folder would contain files with definitions of variables related to individual hosts. Variables can define anything of interest for the specific environment: TCP/UDP port numbers, custom proxy configurations, timer values, and so on. Inventory files are created in either INI or YAML format. INI is a file format commonly used for configuration files of software platforms. It was extensively used in older versions of Microsoft Windows to configure the operating system. INI files are simple text files composed of basic structures such as sections, properties, and values.
- **Module:** Ansible modules are units of parameterized code that get executed by Ansible. Every module in Ansible has a specific use. There are modules, for example, for installing server packages, for managing users, for configuring NTP servers on Cisco NX-OS switches, and for performing **show** commands on Cisco IOS devices. An individual module is invoked through an Ansible task, or multiple modules can be invoked through an Ansible playbook.

Ansible can be installed in several different ways:

- **Using the operating system package manager:** For example, on Fedora, you can install it with the following command:

```
sudo dnf install ansible
```

- **Using the Python package manager:** The command for installing using the Python package manager is **pip install ansible**. As always, it is recommended to use a virtual environment when working with Python. Some dependencies might need to be installed before you can install Ansible. Always check the latest release notes at https://docs.ansible.com.
- **Using the development version:** You can clone the developer repository and issue the **source** command from a bash terminal.

When automating server configurations, Ansible uses SSH to log in to the server, copies the Python code, and runs that code on the server. There is no need to have any Ansible component installed on the managed servers, since the implementation is agentless. For network devices, because there is limited Python support on the devices themselves, the Python code that represents the Ansible jobs runs locally on the control host. The control host still uses SSH or NETCONF, RESTCONF, SNMP, and other interfaces to connect to the network devices and perform the desired configuration changes. Ansible has a large number of modules that support several Cisco operating systems, including IOS, IOS XE, IOS XR, and NX-OS. Some of the most popular Ansible modules that are used with Cisco devices are the following:

- **cisco.ios.ios_command** to send mostly **show** commands to devices running IOS and IOS XE operating systems
- **cisco.ios.ios_config** to send configuration commands to IOS and IOS XE devices
- **cisco.nxos.nxos_command** and **cisco.nxos.nxos_config** to interact with devices running the NX-OS operating system

Next, let's explore a sample Ansible project. Two files are required to get started with Ansible playbook projects: an inventory file and a playbook. In the following Ansible project, the inventory file is called **hosts**, and the playbook file is called **site.yml**:

```
$ tree
├── hosts
└── site.yml


0 directories, 2 files
```

The hosts file contains an inventory of all the devices that will be managed by Ansible. For example, the hosts file can look like this:

```
$ cat hosts

[iosxe]

10.10.30.171



[iosxe:vars]

ansible_network_os=cisco.ios.ios

ansible_connection=network_cli
```

The brackets are used to define group names. Groups are used to classify hosts that share a common characteristic, such as function, operating system, or location. In this case, the [iosxe] group of devices contains only one entry: the management IP address of a Cisco Catalyst 8000v router. The **vars** keyword is used to define variables. In this example, two variables are defined for the iosxe group. The **ansible_network_os** variable specifies that the type of operating system for this group of devices is IOS, and the **ansible_connection** variable specifies that Ansible should connect to the devices in this group by using network_cli, which means SSH. Variables can be referenced in playbooks; as mentioned previously, as the inventory files become larger, it is a good idea to separate the variables' definitions into separate files.

The site.yml file in this example uses the **cisco.ios.ios_command** Ansible module to send two commands to the iosxe group of devices: **show version** and **show ip interface brief**. The YAML definition of the playbook looks as shown in Example 15-1.

Example 15-1 *Ansible Playbook Example*

```
$ cat site.yml
---
- name: Test Ansible ios_command on Cisco IOS XE
  hosts: iosxe

  tasks:
    - name: show version and ip interface brief
      cisco.ios.ios_command:
        commands:
            - show version
            - show ip interface brief
```

YAML documents can start with **---** (and so do Ansible playbooks). A playbook contains one or multiple plays. The plays are logical groupings of tasks and modules. The playbook in this example contains only one play that is marked by the hyphen **(-)** character at the leftmost position. The name of the play is optional, but it should contain an arbitrary string describing the actions that the play will perform. The name is displayed on the screen when the playbook is executed. The **hosts** keyword specifies which hosts or machines the play will be executed against. In Example 15-1, the play is executed against the iosxe group of devices. The value for the **hosts** parameter in the playbook must match the names of the groups, as defined in the inventory file. Tasks are executed on the hosts that are defined in the play definition. For each task, a name value should contain an arbitrary description of the task. Like the name of the play, the name of the task is optional, but it should contain pertinent description text, because it is displayed to the screen when the playbook is executed. **cisco.ios.ios_command** is the name of the Ansible module that is part of the task in this example. The **cisco.ios.ios_command** module takes in multiple parameters, but only one of them is mandatory: the actual commands that will be sent to the IOS device. The two commands that will be sent to the [iosxe] device are **show version** and **show ip interface brief**.

Some of the other optional parameters for the **cisco.ios.ios_command** module are the following:

- **interval:** Specifies the interval of time, in seconds, to wait between retries of the command.
- **retries:** Configures the number of retries for a command before it is considered failed.
- **wait_for:** Specifies a list of conditions that have to be evaluated against the output of the command. In this example, it could be defined as **wait_for: result[0] contains IOS-XE**, which verifies that the output of the **show version** command contains the value **IOS-XE** before going further with the execution of the playbook.

The playbook can be run from a terminal with the following command:

```
$ ansible-playbook -i hosts site.yml -u admin -k
```

The **-i** option specifies the name of the inventory file, the **-u** option specifies the username that will be used to connect to the device, and the **-k** option specifies that the user should be asked for the connection password when the playbook is executed. Connection credentials can also be included in the inventory file, but in this example they are passed in as parameters of the **ansible-playbook** command. The output of this command looks like that shown in Example 15-2.

Example 15-2 *Output of* ***ansible-playbook*** *Command*

```
$ ansible-playbook -i hosts site.yml -u admin -k
SSH password:

PLAY [Test Ansible ios_command on Cisco IOS XE]
**********************************
****************************************

TASK [show version and ip interface brief]
****************************************
**********************************
ok: [10.10.30.171]

PLAY RECAP
********************************************************************************
*****
10.10.30.171               : ok=1    changed=0    unreachable=0    failed=0
                                skipped=0 rescued=0 ignored=0
```

The names of the play and the task are displayed to the screen, as are a play recap and color-coded output based on the status of the playbook execution. In this case, the playbook ran successfully, as indicated by the value **1** for the **ok** status. You can reuse the output of the two **show** commands in the playbook to build custom automation logic, or you can inspect the module return values by using the **-v** option with the **ansible-playbook** command.

Terraform

Terraform is an infrastructure as code tool by HashiCorp. It enables you to define, provision, and manage infrastructure using declarative configuration files written in the Terraform language, which uses HCL syntax. Instead of manually configuring devices or clicking through GUIs, you describe the desired state of infrastructure, and Terraform figures out the steps to reach that state (create/update/delete) in a repeatable way. For Cisco solutions, Terraform can manage configurations by using a provider that talks to the devices or platforms via NETCONF, RESTCONF, REST API, and so on. This capability enables consistent configuration, version control, and automated deployment across many devices and platforms. HashiCorp distributes Terraform as an executable CLI that you can install on supported operating systems, including Microsoft Windows, macOS, and several Linux distributions. You can also compile the Terraform CLI from source if a precompiled binary is not available for your system. For more details on installation, see https://developer.hashicorp.com/terraform.

With Terraform, you declare what the configuration should look like and not the step-by-step CLI commands and how to create it. Ansible playbooks run ordered tasks specified in YAML files, whereas Terraform uses a declarative model in which the desired state is specified in .tf files. Terraform computes a plan (diff) between what exists now (from the device/provider) and what the configuration declares.

A Terraform provider is a plugin that knows how to talk to a specific platform. Terraform Core automatically calculates dependencies between resources to create or destroy in the correct order. Thousands of providers in the Terraform registry allow you to manage your infrastructure as code.

A Terraform resource is a manageable object such as an interface configuration, VLANs, VRFs, or routing protocol. A simple example of an IOS XE loopback interface resource definition in Terraform looks like:

```
resource "iosxe_interface_loopback" "loopback100" {

                name                = 100

                description.        = "Management loopback"

                shutdown.           = false

}
```

A Terraform data source allows you to fetch information from an external system without creating or managing that resource. Think of a data source as a read-only query that retrieves existing data to use in your configuration. An example of a data source that reads the interface configuration for loopback interface 200 and stores it for reference in the loopback200 variable is as follows:

```
data "iosxe_interface_loopback" "loopback200" {

                name                = 200

}
```

While Terraform resources are used to create, update, or delete infrastructure, data sources are used to read and reference existing infrastructure. Data sources are used to avoid hardcoding IDs, IPs, names, and so on by querying them dynamically and using them to cross-reference between resources.

Terraform also maintains a state file (terraform.tfstate) that tracks what it believes exists. By refreshing remote object data through providers and comparing the configuration with state, Terraform calculates what it needs to do to accomplish the desired state and therefore avoid redoing work. Data source results are recorded in terraform.tfstate, but Terraform does not manage the referenced objects; depending on their inputs, reads can occur during refresh or planning or be deferred until they apply.

A Terraform module is a container for multiple resources that are used together. A module is the primary way to organize, package, and reuse Terraform configurations. Think of it as a function in programming: It takes inputs (variables), does work (creates resources), and returns outputs. Any directory containing .tf files is technically a module. You define the configuration pattern (for example, three VLANs, loopback interface, NTP servers, OSPF configuration) as a module that you can call as many times as you want with different inputs (such as site name or IPs). Modules have several key benefits, including the following:

- **Reusability:** Write once, use many times.
- **Organization:** Break large configurations into logical chunks.
- **Consistency:** Use the same pattern across all instances, reducing human error.
- **Abstraction:** Hide complexity behind a simple interface.
- **Collaboration:** Enable different teams to share modules.
- **Versioning:** Pin configurations to tested module versions.

A module could have a directory structure similar to the following:

modules/

```
└── test-module/

    ├── main.tf        # Resources

    ├── variables.tf   # Input variables

    ├── outputs.tf     # Output values

    └── README.md      # Documentation
```

A simple main.tf example file that configures a loopback 100 interface on a Cisco IOS XE device could look like the following. The Cisco IOS XE provider uses NETCONF over SSH, so NETCONF must be enabled on the device (for example, with netconf-yang):

```
terraform {

    required_providers {

        iosxe = {

            source  = "CiscoDevNet/iosxe"

            version = "~> 0.18"

        }

    }

}


provider "iosxe" {

  # Set IOSXE_HOST in the environment.

  # Set IOSXE_USERNAME in the environment.

  # Set IOSXE_PASSWORD in the environment.

}


# Loopback interface configuration

resource "iosxe_interface_loopback" "lo0" {

  name                            = 100

  description                          = "Managed by Terraform"

  shutdown                         = false

  ipv4_address                       = "10.10.100.100"

  ipv4_address_mask           = "255.255.255.255"

}
```

The Terraform configuration lifecycle contains several steps: init > plan > apply. These steps are controlled by the following CLI commands:

- **terraform init:** This command triggers the download of the required providers specified in the Terraform configuration and sets up the backend.
- **terraform plan:** This command shows the intended changes of the configuration, but it does so in a safe preview mode or dry run mode without actually applying the configuration to the devices.
- **terraform apply:** This command applies the planned configuration changes on the target devices.
- **terraform destroy:** This command removes what Terraform created.

Terraform brings declarative, version-controlled, plan/apply workflow to network automation. It excels at defining and enforcing desired network state, configuration drift detection, and remediation and scaling standard configs across many devices and sites.

Cisco Network Services Orchestrator (NSO)

Cisco *Network Services Orchestrator (NSO)* is a network services orchestration platform that enables end-to-end service deployment across multivendor physical and virtual infrastructure. NSO can also be considered a multivendor service-layer SDN controller for data center, enterprise, and service provider networks. It takes full advantage of NETCONF and YANG data models to provide a single API and a single user interface to the network that it manages. Cisco NSO is the single source of truth in a network, maintaining service and device configuration in its configuration database. Devices can drift from CDB; operators use check-sync and sync-from or sync-to to detect and reconcile differences.

The main components of Cisco NSO are as follows (see Figure 15-4):

- Service manager
- Device manager
- Mapping logic
- Configuration database

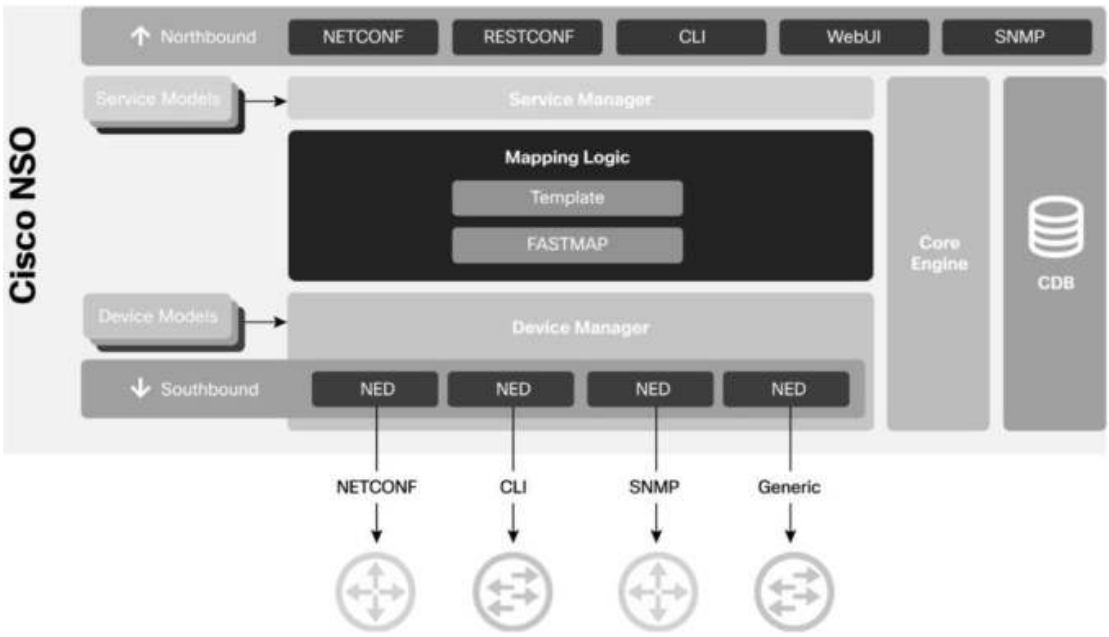


Figure 15-4 *Cisco NSO Architecture*

The two primary technologies used with Cisco NSO are the following:

- **NETCONF:** Used for standard and efficient automation of configuration
- **YANG:** Used for services and device configuration data modeling

One of the main advantages of Cisco NSO is the model-to-model mapping capability. A network administrator can define the data models for the end-to-end services that need to be implemented in the network (for example, Layer 3 MPLS VPNs, IPTV), and NSO maps those service models into device models for various vendor devices. Because a large number of devices do not fully support NETCONF and YANG data models, NSO uses Network Element Drivers (NEDs) to model native device CLIs into YANG models. NSO offers a level of abstraction that makes network devices transparent to the service management of the network. This way, complex services can be described and implemented in simple ways and pushed to the devices no matter what the device vendor, configuration semantics, and so on.

NSO provides several northbound interfaces for service and platform management, including NETCONF, RESTCONF, JSON-RPC, a CLI, a web user interface, and SNMP. The NSO CLI and web user interface are usually used for human interaction with the platform; SNMP and NETCONF are used for monitoring; and the NETCONF, RESTCONF, and JSON-RPC interfaces are used for automation integration. The NSO CLI has two modes of operation: operational mode, used primarily to monitor and maintain the platform, and configuration mode, used to configure the services, the devices, and the NSO server. Two CLI styles are offered to NSO operators: Cisco style, which is identical to the Cisco CLI, and Juniper style, which is identical to the Juniper CLI. CLI users can easily switch between them by using the **switch cli** command.

The southbound communication with the network devices is done through components called Network Element Drivers, which implement a large set of southbound interfaces, including NETCONF, the CLI, SNMP, and OpenFlow.

The logical architecture of Cisco NSO is a dual-layered approach: a device manager component that handles the device configuration scenarios and a service manager component that provides an interface for the administrator to define the services that need to be implemented in the network. The configuration data store, called the *Configuration Database (CDB)*, is in sync with all the devices in the network and contains all the devices and services configurations. A dedicated mapping layer manages the correspondence between the service models and the device models.

The Cisco NSO Core Engine manages critical operational functions such as transactions, high-availability replication, upgrades and downgrades, role-based access control, and rollback management. All operations in NSO are handled by the transaction manager. The Core Engine connects all the other NSO components together; it is the communication backbone for all other components and is the process that reads the initial configuration defined in the ncs.conf file.

The CDB stores all the platform data in a RAM database, including the NSO configuration, the configurations of all services and all managed devices, and all NSO operational data. The CDB is stored in RAM for increased speed and quick response, and a persistent version of it is stored on the disk drive of the server. The CDB is a hierarchical database organized in a tree-like structure and provides NSO's built-in transactional configuration store.

With the Cisco NSO service manager, users develop YANG models of the services they want deployed in the network, such as IPTV or MPLS VPNs. At this stage, the service model also has to be mapped to the corresponding device configuration model. This is done either through configuration templates that map service parameters into device configuration parameters or programmatically by building the mapping logic with Java or Python code. The lifecycle of the services defined in NSO is managed by a FASTMAP algorithm. As services get created, modified, and deleted, FASTMAP dynamically reacts and adjusts the configuration of the devices in the network to reflect the status of the service models. In addition, FASTMAP enables the following functions:

- **Service dry-run:** NSO calculates what the device changes would be if the service model were to be pushed on the devices in the network.
- **Service check-sync:** The service configuration is verified to be in sync with the actual device configuration. Device configuration drift can be detected this way.
- **Service re-deploy:** Device configurations can be redeployed to restore a service on the network.

The NSO device manager manages the network devices by using YANG data models and NETCONF. For devices that natively implement NETCONF and YANG models, the device manager is automatic; devices that do not support NETCONF are integrated in the platform with NEDs. The installer includes sample NEDs for evaluation; production NED packages for specific platforms are obtained separately. If needed, custom NEDs can be developed to address any device management protocol that is not already included with the default installation. Distributed atomic transactions are used for all configuration changes on all the devices in the network. If applying the configuration changes fails on any of the devices in the service path, a rollback mechanism can be used to revert all the changes and return the network devices to their initial status before the service deployment was attempted. Configuration templates can be used at this stage to easily and quickly provision new devices.

As of this writing, an evaluation version and current installation instructions are available in the Cisco NSO Quick Start Guide at https://nso-docs.cisco.com/quick-start/quick-start-guide. The following examples use Cisco NSO version 5.1 running locally on macOS Catalina.

Cisco NSO provides a network simulation tool called **ncs-netsim**. This tool makes it easy to test NSO packages against simulated devices and to learn how to use the platform without any hardware needed. The same YANG models are used for both real and NetSim simulated devices. Several options are available with ncs-netsim, as shown in Example 15-3.

Example 15-3 *ncs-netsim Options*

```
$ ncs-netsim --help
Usage ncs-netsim  [--dir <NetsimDir>]
                  create-network <NcsPackage> <NumDevices> <Prefix> |
                  create-device <NcsPackage> <DeviceName>           |
                  add-to-network <NcsPackage> <NumDevices> <Prefix> |
                  add-device <NcsPackage> <DeviceName> |
                  delete-network                       |
                  [-a | --async]  start [devname]      |
                  [-a | --async ] stop [devname]       |
                  [-a | --async ] reset [devname]      |
                  [-a | --async ] restart [devname]    |
                  list                         |
                  is-alive [devname]           |
                  status [devname]             |
                  whichdir                     |
                  ncs-xml-init [devname]       |
                  ncs-xml-init-remote <RemoteNodeName> [devname] |
                  [--force-generic]                    |
                  packages                     |
                  netconf-console devname [XpathFilter] |
                  [-w | --window] [cli | cli-c | cli-i] devname |
                  get-port devname [ipc | netconf | cli | snmp]
See manpage for ncs-netsim for more info. NetsimDir is optional and defaults to
./netsim, any netsim directory above in the path, or $NETSIM_DIR if set.
```

To create a new network simulation, you use the **create-network** option with the following parameters:

```
ncs-netsim create-network <NCS package> <#N devices> <PrefixY>
```

<NCS package> contains the path to where the NCS packages were installed. By default, this is $NCS_DIR/packages/neds, and it contains NCS packages for Cisco IOS, Cisco IOS XR, Cisco NX-OS, Juniper JunOS, and so on. The *<#N devices>* parameter specifies the number of devices that will be created, and *<PrefixY>* defines the prefix that will be used to name the devices. For example, the following command would be used to create a network simulation with three Cisco IOS devices named ios0, ios1, and ios2:

```
$ ncs-netsim create-network $NCS_DIR/packages/neds/cisco- ios-cli-3.8 3 ios

DEVICE ios0 CREATED

DEVICE ios1 CREATED

DEVICE ios2 CREATED
```

After the virtual devices are created, the simulation can be started with the **start** command:

```
$ ncs-netsim start

DEVICE ios0 OK STARTED

DEVICE ios1 OK STARTED

DEVICE ios2 OK STARTED
```

An ordered list of all the devices that are running at any point on the NSO server can be obtained by using the **list** option for ncs-netsim, as shown in the following output:

```
$ ncs-netsim list
ncs-netsim list for $NCS_DIR/nso-run/netsim

name=ios0 netconf=12022 snmp=11022 ipc=5010 cli=10022 dir=$NCS_DIR/
  nso-run/netsim/ios/ios0
name=ios1 netconf=12023 snmp=11023 ipc=5011 cli=10023 dir=$NCS_DIR
  /nso-run/netsim/ios/ios1
name=ios2 netconf=12024 snmp=11024 ipc=5012 cli=10024 dir=$NCS_DIR
  /nso-run/netsim/ios/ios2
```

Access to the console ports of all the simulated devices is possible by using the **cli-i** option followed by the name of the device, as in Example 15-4, which demonstrates console access to the ios0 device.

Example 15-4 *CLI Access to a Simulated Device*

```
$ ncs-netsim cli-i ios0

admin connected from 127.0.0.1 using console on *
ios0> en
ios0# show running-config
no service pad
no ip domain-lookup
no ip http server
no ip http secure-server
ip routing
ip source-route
ip vrf my-forward
 bgp next-hop Loopback 1
!
```

The simulated devices implement all control plane functions of the original operating system, but they do not have the data plane implemented, so they do not forward data traffic. Even with this limitation, ncs-netsim network simulations are used extensively for testing and learning.

As mentioned previously, Cisco NSO exposes a RESTCONF northbound interface for automation and integration purposes. Before simulated devices are accessible over this interface, they have to be onboarded within Cisco NSO. Devices can be onboarded through any of the northbound interfaces that Cisco NSO exposes, but in this example, the NSO CLI was chosen. First, access to the NSO Cisco version (**-C** parameter) of the CLI for the admin user is obtained by issuing the **ncs_cli -C -u admin** command. The default password is also **admin**. Once access to the NSO CLI is granted, all the simulated Cisco IOS devices need to be added in the configuration of the Cisco NSO server. Example 15-5 shows the configuration for ios0.

Example 15-5 *Onboarding Simulated Devices in Cisco NSO*

```
$ ncs_cli -C -u admin

admin connected from 127.0.0.1 using console on *
admin@ncs# config term
Entering configuration mode terminal
admin@ncs(config)# devices device ios0
admin@ncs(config-device-ios0)# address 127.0.0.1
admin@ncs(config-device-ios0)# port 10022
admin@ncs(config-device-ios0)# authgroup default
admin@ncs(config-device-ios0)# device-type cli ned-id cisco-ios-cli-3.8
admin@ncs(config-device-ios0)# state admin-state unlocked
admin@ncs(config-device-ios0)# exit
```

When all three simulated IOS devices are onboarded and the configuration changes are committed, the devices can be displayed through a GET call over the RESTCONF interface. Cisco NSO exposes its full functionality over this RESTCONF interface that is available by default starting at **http://<NSO_Server_IP>:8080/restconf/ root**. A list of all the devices that have been onboarded can be obtained by performing a GET call on the following RESTCONF URI resource: http://<NSO_Server_IP>:8080/restconf/data/tailf-ncs:devices/device. The **curl** command to test this GET call looks as follows:

```
$ curl --request GET 'http://localhost:8080/restconf/data/tailf-ncs:devices/device' /

--header 'Accept: application/yang-data+json' /

--header 'Authorization: Basic YWRtaW46YWRtaW4='
```

The Accept header requests **application/yang-data+json** as the response media type for JSON-encoded YANG data. The authorization header contains the basic authentication Base64 encoded for the **admin** username and the **admin** password. A snippet of the response is shown in Example 15-6.

Example 15-6 *Output of the RESTCONF GET Devices API Call*

```
{
  "tailf-ncs:device": [
    {
      "name": "ios0",
      "address": "127.0.0.1",
      "port": 10022,
      "authgroup": "default",
      "device-type": {
       "cli": {
          "ned-id": "cisco-ios-cli-3.8:cisco-ios-cli-3.8"
        }
      },
      "commit-queue": {
        "queue-length": 0
      },
      "active-settings": {
        "connect-timeout": 20,
        "read-timeout": 20,
        "write-timeout": 20,
        "ssh-keep-alive": {
          "interval": 20,
          "count": 3
        },
        "ned-keep-alive": {
          "count": 3
        },
... omitted output
}
```

The Cisco NSO RESTCONF interface can also be explored by using Postman. For example, you can get the configuration of the ios1 simulated device from the NSO RESTCONF interface by using Postman instead of **curl**. The URI that returns this information is similar to the one explored previously, with one difference: http://localhost:8080/restconf/data/tailf-ncs:devices/device=ios1. device=ios1 is used to indicate that the API should return only data specific to the device ios1. The Accept header value application/yang-data+json and Basic Auth credentials with username **admin** and password **admin** need to be included. Figure 15-5 shows the response from the NSO server after the Send button is clicked.

Figure 15-5 *RESTCONF GET Device API Call in Postman*

Cisco Modeling Labs (CML)

For a long time, the process for building and maintaining infrastructure laboratories for testing and learning was time-consuming and required a lot of hardware, which needed space, power, and cooling; these labs were therefore very expensive and within reach of only a few companies. With the advent of Network Function Virtualization (NFV) and *Cisco Modeling Labs (CML)*, it has finally become easy and cost-effective to build infrastructure laboratories for all types of purposes.

NFV is meant to decouple the hardware and software requirements that are typical for network devices and enable the software component or the network operating system to run on commodity hardware as a virtual machine, in a container, or on bare-metal servers. While, in most cases, this effort leads to a loss of performance because custom hardware components such as application-specific integrated circuits (ASICs) are not used with NFV offerings, it is usually considered an acceptable trade-off and offers a cost-effective way to take advantage of the software features that come with the network operating system. If maximum performance is required, a combination of dedicated custom hardware and software can be used to ensure guaranteed performance. If the performance requirement is not as stringent, an NFV alternative that includes just the software network operating system running on off-the-shelf hardware can be considered. For example, Cisco offers the enterprise-grade network operating system Cisco IOS XE as a downloadable virtual machine in the form of a Cisco Catalyst 8000v router for an NFV offering covering enterprise routing capabilities, and it also offers IOS XE as a combination of hardware and software with the Cisco Catalyst 9000 Series of switches that offer 100 Gbps interfaces and guaranteed hardware routing and switching forwarding performance. There is a virtual option for almost all Cisco operating systems currently available, including Cisco IOS XE with Cisco Catalyst 8000v, Cisco IOS XR with Cisco IOS XRv, and Cisco NX-OS with Cisco Nexus 9000v. These virtual instances can be run on x86 commodity hardware on premises or in private and public cloud environments.

Cisco CML is a powerful network simulation platform that is meant to be a flexible, all-in-one virtual networking lab. With Cisco CML and virtual instances of network operating systems from Cisco, it is possible to build entire network simulations. The CML version 2 web interface looks as shown in Figure 15-6.

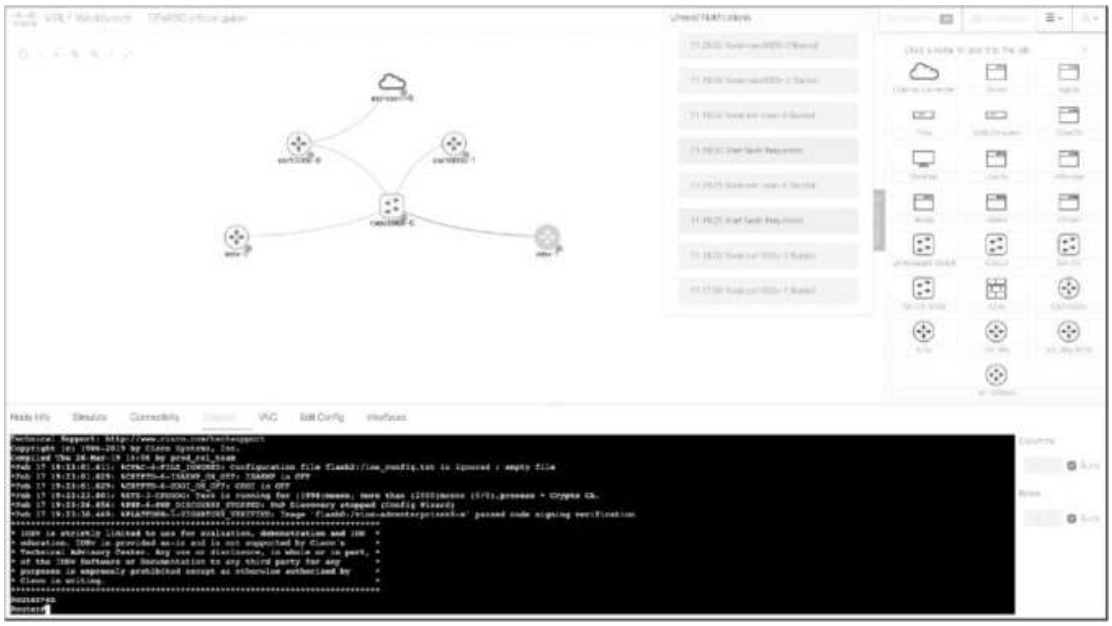


Figure 15-6 *CML 2 Web Interface*

Cisco CML provides several interfaces: a CLI interface, a powerful web user interface (refer to Figure 15-6), and a REST API for automation and integration. Multiple virtual instances of operating systems and solutions—including ASAv, Catalyst 8000V, Catalyst 9000v, CSR 1000v, IOSv, IOSvL2, IOS XRv 9000, Nexus 9300v, Alpine Linux, Ubuntu, and TRex—are supplied in the CML reference platform ISO, and it is possible to add other third-party virtual instances as well. While the software features in the virtual instances of the Cisco network operating systems are, in most cases, on parity with the binary files installed on Cisco routers and switches, there are limitations in regard to the data plane implementation. There are no switching backplanes and no ASIC modules with the virtual nodes in CML, so throughput performance cannot be tested accurately in most cases.

Using an external connector, a simulation running in Cisco CML can be connected to the external world and the Internet. This is done through the network interface on the server where Cisco CML is installed. Cisco CML can be deployed as an OVA on supported VMware hypervisors; bare-metal deployments use the controller ISO.

Several lab simulations can be run at the same time. Going back to the web user interface from Figure 15-6, under the Lab Manager tab, you can create, modify, or delete new labs. A lab in CML is a logical construct used to organize and separate network topologies into separate environments. Once a lab is created, a network topology can be built with the nodes and virtual

instances that are available in CML. Network interfaces can be dynamically added and removed from CML nodes and can be easily interconnected by simply hovering the mouse pointer over the devices, selecting the network connection option, and then dragging the virtual network cable between the devices that need to be connected. Direct access to the console of the nodes is also managed through CML. In the case of servers, VNC connections are also supported. Startup configurations for network devices can be specified prior to starting a simulation. Network topologies are stored as YAML files and can be easily modified and shared. Example 15-7 shows a snippet of a CML network topology file.

Example 15-7 *CML Topology File Example*

```
lab:
  description: ''
  notes: ''
  timestamp: 1581966586.8872395
  title: DEVASC official guide
  version: 0.0.3
nodes:
  - id: n0
    label: iosv-0
    node_definition: iosv
    x: -500
    y: 50
    configuration: ''
    image_definition: iosv-158-3
    tags: []
    interfaces:
      - id: i0
        label: Loopback0
        type: loopback
      - id: i1
        slot: 0
        label: GigabitEthernet0/0
        type: physical
  - id: n1
    label: csr1000v-0
    node_definition: csr1000v
...omitted output
```

The procedure for downloading and uploading CML network topology YAML files is easy and straightforward, and CML users can easily share topologies and also store them in version control systems such as Git. CML can be part of a CI/CD pipeline for network configuration changes. An example of such a pipeline could contain the following components and processes:

- CML topology files could be stored together with automation scripts, device configuration files, and possibly Ansible playbooks in a version control system.
- Whenever a change is made to any of these files, a build automation tool such as Jenkins or Drone can be used to monitor and detect the change in version and initiate the automation processes.
- A complete automated process can create a test environment using the CML REST APIs, build application servers, and deploy them in the test environment.
- The network configuration changes can be tested in this CML virtual environment and a pass or fail criterion can be set based on the expectations of the administrator.
- If the tests pass, a configuration automation solution like Ansible could use playbooks and automation scripts to apply the tested configuration changes in the production network.

Python Automated Test System (pyATS)

The *Python Automated Test System (pyATS)* and the pyATS library together form the Cisco test and automation solution, a vibrant ecosystem that aims to standardize how automated network tests are set up and run. pyATS was developed by Cisco in 2014 and is used extensively internally by thousands of engineers to run unit tests, regression tests, and end-to-end and integration tests for a large number of Cisco products. The solution is completely developed in Python 3, making it easy to work with, scalable, and extensible. Millions of pyATS tests are run every month internally at Cisco. pyATS tests provide sanity, feature, solution, system, and scale checks for any type of physical or virtual device, including routers, switches, access points, and firewalls.

The solution has two main components: the pyATS test framework and the pyATS library, which used to be called Genie but was renamed in an effort to simplify the nomenclature of the product. The pyATS test framework provides ways to define how the network topologies are created and modeled, how to connect to devices through connection libraries, and how to actually perform the tests and generate reports. The pyATS library builds on this infrastructure framework and provides easy-to-use libraries that implement pyATS features, parsers to interpret the data received from the devices, a mechanism for modeling network device configurations for both Cisco and third-party vendors, reusable test cases in the form of triggers and verifications, and the ability to build test suites through YAML-based text files. In the rest of this chapter, when pyATS is mentioned, assume that we are referring to the product as a whole, unless otherwise specified. Figure 15-7 shows the components of the pyATS solution.

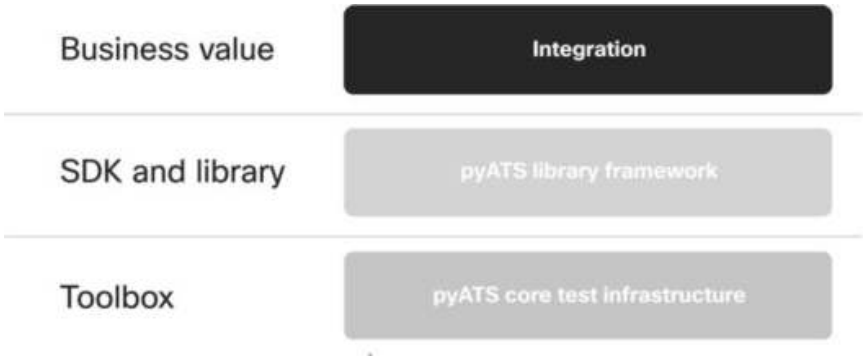


Figure 15-7 *pyATS Architecture*

pyATS fits well in infrastructure automation use cases, especially around configuration change validation. As discussed earlier in this chapter, tools like Ansible and Terraform are great at infrastructure configuration automation, and pyATS complements them with operational pre-change and post-change validation. pyATS can be easily integrated in day-to-day DevOps activities and CI/CD automation pipelines. Some of the most common use cases for pyATS are as follows:

- **Profiling the current status of a network and taking a snapshot of both the configuration status as well as the operational data of the network:** This can be done before and after a configuration change or a software upgrade/downgrade is performed to ensure that the network still performs within desired parameters. For example, a snapshot of the network is taken with pyATS before a software upgrade is performed, and key metrics are noted, such as number of BGP sessions in the established state or the number and type of routing table entries or any other metric that is considered critical for that environment. The software upgrade is completed, and then a new pyATS snapshot is taken to ensure that those critical metrics have values within expected parameters. pyATS offers all the tooling to be able to automatically perform the network snapshots, compare key metric values, set pass/fail criteria, and even generate reports.
- **Automating configuration changes and monitoring of devices:** By using the Conf module within the pyATS library, configuration changes can be performed in a uniform fashion across different devices from different vendors. Together with the profiling capabilities, pyATS offers a full set of capabilities for infrastructure automation, including creating a snapshot of the network pre-change, performing the changes, taking another snapshot post-change, and continuously monitoring for any performance outliers.

pyATS offers an abstraction layer and programmatically stores device configuration and operational data in Python objects such as dictionaries and lists. It is not necessary to screen scrape and parse different proprietary CLI outputs, because pyATS automatically accomplishes these tasks through its extensive collection of parsers and outputs the data into standard formats that can be easily consumed. The platform is easily extensible and customizable with scripts and test cases that pertain to various environments. A pyATS CLI is also available for network testers who are not comfortable with Python. pyATS supports several options to connect to devices: Telnet, SSH, REST, NETCONF, and RESTCONF.

pyATS can be installed through **pip** by issuing the following command:

```
pip install pyats[full]
```

Different options are available:

- **pyats[full]** installs all pyATS components, the pyATS library framework, and optional extras.
- **pyats[library]** installs pyATS core plus the pyATS Library (Genie).
- **pyats** without any options installs just the pyATS test infrastructure framework.

pyATS releases are published frequently for supported Python environments on Linux and macOS. Microsoft Windows is not directly supported, but Windows Subsystem for Linux (WSL) can be used to run pyATS on Windows. As usual, it is recommended to install pyATS in its own virtual environment. You can find a prebuilt Docker container with pyATS already

installed at https://hub.docker.com/r/ciscotestautomation/pyats/. You also can find more up-to-date information on pyATS versions, documentation, and installation steps at https://developer.cisco.com/pyats/.

To start working with pyATS, you need to create a testbed YAML file. This file contains the connectivity details for all the devices in the testbed. Example 15-8 shows a YAML file with only one device.

Example 15-8 *pyATS Testbed File*

```
---
devices:
catalyst8000v-1:
     type: 'router'
     os: 'iosxe'
     platform: asr1k
     alias: 'uut'
     credentials:
         default:
             username: vagrant
             password: vagrant
     connections:
         cli:
             protocol: ssh
             port: 2222
             ip: '127.0.0.1'
```

Most of the options, while defining devices in a testbed, should be self-explanatory. Just a few options need to be explained in more detail. First, the hostname—which is catalyst8000v-1 in Example 15-8—has to match the configured hostname of the device. Second, the alias is used to identify the device during script execution. The test script can be reused on a different testbed as long as the aliases stay the same. A complete list of values for the type, operating system, and platform is available with the pyATS documentation. Say that you save this file and call it testbed.yaml.

Example 15-9 shows a simple Python script that uses the pyATS library and loads the testbed.yaml file, searches in that file for a device with alias **uut**, connects to that device, runs the **show version** command, and extracts the Cisco IOS XE version number and prints it to the screen.

Example 15-9 *pyATS Test*

```
#! /usr/bin/env python3
from genie.testbed import load

# Load the testbed
tb = load('testbed.yaml')

# Find the device with alias uut
dev = tb.devices['uut']

# Connect to the device
dev.connect()

# Parse the output of the show version command
output = dev.parse('show version')

# Extract the version number and print it to the screen
print('IOS-XE version number: ' + output['version']['version'])
```

The **output** variable is a Python dictionary that contains structured data parsed from the output of the **show version** command. This makes it easy to parse and extract the needed data from the output of the **show** command.

Exam Preparation Tasks

As mentioned in the section "How to Use This Book" in the Introduction, you have a couple of choices for exam preparation: the exercises here, Chapter 19, "Final Preparation," and the exam simulation questions on the companion website.

Review All Key Topics

Review the most important topics in this chapter, noted with the Key Topic icon in the outer margin of the page. Table 15-3 lists these key topics and the page number on which each is found.

Table 15-3 *Key Topics*

Key Topic Element	Description	Page Number
Paragraph	Network controllers	
Paragraph	Direct device configuration	
Paragraph	Infrastructure as code	
List	Two approaches to infrastructure as code: declarative and imperative	
Paragraph	Continuous integration/continuous delivery	
Paragraph	Build server	
Paragraph	CI/CD pipelines	
List	Ansible concepts	
Paragraph	Sample Ansible project	
Paragraph	Ansible playbooks	
Paragraph	Terraform	
List	Main components of Cisco NSO	
Paragraph	Logical architecture of Cisco NSO	
Paragraph	GET call over the RESTCONF interface	
Paragraph	Cisco CML	
Paragraph	pyATS test framework and pyATS library	
Paragraph	Testbed YAML file for pyATS	

Define Key Terms

Define the following key terms from this chapter and check your answers in the glossary:

continuous integration/continuous delivery (CI/CD) pipeline
Network Services Orchestrator (NSO)
Configuration Database (CDB)
Cisco Modeling Labs (CML)
Python Automated Test System (pyATS)

Chapter 16. Network Fundamentals

This chapter covers the following topics:

- **Network Reference Models:** This section covers the two most popular networking reference models: the OSI model and the TCP/IP model.
- **Switching Concepts:** This section introduces Layer 2 switching concepts such as Ethernet, MAC addresses, and VLANs.
- **Routing Concepts:** This section delves into Layer 3 routing concepts and introduces IP addressing.

This chapter covers network fundamentals. It starts with a short introduction to what networking is and how it has evolved over the years. It also takes an in-depth look at the two most popular networking reference models: the OSI model and the TCP/IP model. This chapter also explores critical Layer 2 technologies, including Ethernet, which has become the de facto protocol for local-area networks (LANs), MAC addresses, virtual local-area network (VLANs), switching, and switching tables. Internet Protocol (IP) is the Layer 3 protocol that the Internet is built on. This chapter explores both IP version 4 and version 6, as well as what routing is, how routing tables are built, and how data packets are forwarded over networks.

"Do I Know This Already?" Quiz

The "Do I Know This Already?" quiz allows you to assess whether you should read this entire chapter thoroughly or jump to the "Exam Preparation Tasks" section. If you are in doubt about your answers to these questions or your own assessment of your knowledge of the topics, read the entire chapter. Table 16-1 lists the major headings in this chapter and their corresponding "Do I Know This Already?" quiz questions. You can find the answers in Appendix A, "Answers to the 'Do I Know This Already?' Quiz Questions."

Table 16-1 *"Do I Know This Already?" Section-to-Question Mapping*

Foundation Topics Section	Questions
Network Reference Models	1–4
Switching Concepts	5–7
Routing Concepts	8–10

Caution

The goal of self-assessment is to gauge your mastery of the topics in this chapter. If you do not know the answer to a question or are only partially sure of the answer, you should mark that question as wrong for purposes of self-assessment. Giving yourself credit for an answer that you correctly guess skews your self-assessment results and might provide you with a false sense of security.

1. Which OSI layer is responsible for establishing an end-to-end connection between the sender and the receiver?
   - **a.** Network layer
   - **b.** Transport layer
   - **c.** Session layer
   - **d.** Presentation layer
2. What layer of the TCP/IP reference model is the equivalent of the network layer in the OSI model?
   - **a.** Physical layer
   - **b.** Data link layer
   - **c.** Internet layer
   - **d.** Transport layer
3. Which of the following are examples of application layer protocols in the TCP/IP reference model? (Choose two.)
   - **a.** TCP
   - **b.** HTTP

c. BGP

d. FTP

4. What is the transport layer PDU called in the TCP/IP reference model?

a. Data

b. Frame

c. Packet

d. Segment

5. What is the role of the Preamble field in the Ethernet header?

a. It is used as padding data to ensure that the frame has at least the minimum number of bytes for transmission.

b. It is used as padding data to ensure that the frame has 1500 bytes for transmission.

c. It is used to ensure that the frame was transmitted without data corruption.

d. It is used to synchronize the signal between the sender and receiver.

6. How many bits are in a MAC address?

a. 24 bits

b. 48 bits

c. 32 bits

d. 64 bits

7. What happens to a data frame for which a switch doesn't have the destination MAC address in its switching table?

a. It gets discarded.

b. It gets transformed into a broadcast data frame.

c. It gets sent back to the sender as it cannot be switched.

d. It gets flooded out all the ports except the port on which it was received.

8. What is the bit pattern in the first byte for Class C IPv4 addresses?

a. 110xxxxx

b. 11110xxx

c. 10xxxxxx

d. 1110xxxx

9. What is the broadcast address for the 192.168.0.96/27 network?

a. 192.168.0.191

b. 192.168.0.159

c. 192.168.0.127

d. 192.168.0.255

10. What are some of the characteristics of IPv6 addresses? (Choose three.)

a. They are 128 bits long.

b. Colons separate 16-bit hexadecimal fields.

c. Hexadecimal characters are case sensitive.

d. Successive fields of zero can be represented as ::.

Foundation Topics

Networks of devices have been built for more than 50 years now. The term *device* is used here to mean any piece of electronic equipment that has a network interface card of some sort. End-user devices or consumer devices such as personal computers, laptops, smart-phones, tablets, and printers as well as infrastructure devices such as switches, routers, firewalls, and load balancers have been interconnected in both private and public networks for many years. What started as islands of disparate devices connected on university campuses in the 1960s and then connected directly with each other on ARPANET has evolved to become the Internet, where everyone and everything is interconnected. This chapter discusses the fundamentals of networking.

Network Reference Models

Several network reference models have been developed through the years, some of them proprietary and some open. Two reference models stand out from the crowd for their importance and longevity: the OSI model and the TCP/IP reference model.

The OSI Model

In the late 1970s, as computing devices became more and more popular, a need to connect them in a manner in which they would be able to exchange data between them arose. The old sneakernet—which involved physically copying data that needed to be transferred between devices on a floppy disk and delivering it on foot—was no longer appropriate for the amounts of data that needed to be exchanged. Disparate proprietary solutions such as System Network Architecture (SNA) from IBM and DECnet from Digital Equipment Corporation (DEC) existed for this purpose, but there was a need to have a standard, interoperable, and common way to do networking. The International Organization for Standardization (ISO) took on this challenge and came up with the Open Systems Interconnection (OSI) model. Although it has not been implemented in production systems, the OSI model is still extensively used especially from an educational perspective. The simpler TCP/IP (Transport Control Protocol/Internet Protocol) model has had much more success in real life and is covered in more detail later in this chapter.

The main idea with developing networking models based on different layers that interact with each other was inspired by the computer science mantra of splitting a difficult problem into smaller more manageable problems. Following this philosophy, the OSI model splits the data transmission processes needed to transfer data between two or more devices into seven layers. An advantage with a layered approach is that there can be different vendors at different layers implementing the specific functionality required by each layer. There are network equipment vendors that focus on developing devices that operate at Layer 2 of the OSI model, called *switches*, and devices that operate at Layer 3 of the model, called *routers*, and so on. But probably the biggest advantage of layering is the level of innovation that this model allows at each layer. As long as the functionality provided by each layer stays consistent with the expectations of the layer above, protocols, hardware, and software can be developed and improved. Take, for example, the way the Ethernet protocol and specifications have evolved over the years from transfer rates of 10 Mbps in the early days of the protocol to 400 Gbps today. This improvement of several orders of magnitude has been possible because of the layering philosophy. As long as the data transmission layer provides a set of primitives to the layer above, it can freely innovate and improve.

In the layered approach, each layer provides a set of functionality and features to the layer above it and consumes the features of the layer below it. The OSI model defines seven layers for exchanging data between networked devices (see Figure 16-1). Two or more devices that would exchange data with each other using the OSI model need to implement all seven layers.

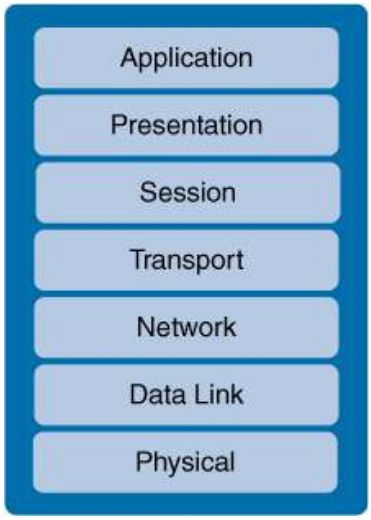


Figure 16-1 *OSI Model Layers*

Applications that need to transmit data over the network start at the application layer, gathering the data that needs to be transmitted. This data makes its way down the transmission model and gets processed at each layer by being split into smaller chunks, with header and footer information added, and then being encoded on the physical medium. As the bits arrive at the destination, the reverse process takes place: As the data moves up the layers at the receiver, headers and footers are removed, and data is recombined into bigger chunks and made available to the receiving application. This is similar in a way to how postal mail works. When sending out heartwarming holiday wishes via snail mail, first you need to put your message on a piece of paper. This is the equivalent of the application layer at the sender.

Then you put the paper with the message in an envelope, seal the envelope, write down the sender and receiver information, and drop the envelope in a mailbox or at the post office. The postal service plays the role of the physical layer in this case, as it sorts

through the mail and delivers each piece to its destination. The person receiving the mail makes sure that the envelope is meant for him or her and then goes ahead and reverses part of the process you went through, removing the paper from the envelope and reading the message. The message is successfully delivered from source to destination in both cases—on a network and via snail mail—but it takes a fraction of a second for data to get anywhere on planet Earth on a network, whereas it might take days or weeks for snail mail to reach a recipient, depending on how far the recipient is from the sender.

Data transmission protocols have been developed at each layer of the OSI model to perform the functionality required by that layer. Each layer is supposed to perform specific tasks and provide specific functionality to the layers above it.

The **physical layer** is tasked with the transmission and reception of raw bit streams. At this layer, specifications about the voltage levels, pinouts for the connectors, physical transmission data rates, modulation schemes, and so on are defined. This is where the proverbial rubber meets the road, and data to be transmitted is converted into electrical, optical, or radio signals. Protocols such as Ethernet and Bluetooth have specifications at the physical layer.

The **data link layer** provides device-to-device data transfer on a local-area network. It defines how data is formatted for transmission and how access to the physical medium is controlled. Errors that might have happened in transit on the physical layer are corrected at the data link layer. Bridges and switches are devices that operate at the data link layer. The IEEE has several protocols defined for the data link layer, organized into the IEEE 802 family of protocols. According to the specifications in these protocols, the data link layer is further subdivided into two sublayers:

- The Media Access Control (MAC) layer provides specifications on how devices gain access to the transmission medium.
- The Logical Link Control (LLC) layer is responsible for encapsulating network layer data and frame synchronization.

The **network layer** provides connectivity between devices that are connected to separate networks. Whereas the data link layer ensures communication between directly connected devices on a local network, the network layer enables communication between devices that are in geographically separate locations. Logical addressing of all the devices on the network and routing of data packets on the best path between a sender and a receiver are also specified at the network layer. Routers are devices that operate at the network layer. If the data that needs to be transmitted over the data link layer is too large to be sent at once, the network layer has the capability to fragment the data into smaller pieces. As the name indicates, the process of *fragmentation* consists of splitting the data into smaller pieces, sending those pieces on the network, and then reassembling them when they arrive at another device on the network. Fragmentation should be avoided as much as possible as it slows down the transmission of data considerably. This slowdown occurs because of how modern routers forward data packets and the mechanism they implement to accomplish this as fast as possible. Most modern routers perform data routing in hardware using specialized integrated circuits called ASICs. When data becomes fragmented, instead of being forwarded in hardware using the ASICs, the packets get process switched, which means they are forwarded based on an interrupt system using the regular CPU on the router. Data delivery at the network layer is best effort, as reliable transmission is not required at this layer. As we'll see shortly, reliable transmission is handled at the next higher layer, the transport layer.

The **transport layer**, as the name suggests, is responsible for end-to-end transport of data from the source to the destination of the data traffic. It segments data that it receives from the layer above and reassembles it into a stream of data on the receiving device. This layer specifies several mechanisms so that data is reliably transmitted between devices. Error detection and recovery mechanisms and information flow control ensure reliable service. This is also where the delineation between the application and the network is usually done. The transport layer and the network, data link, and physical layers make up the data transport layers. The main purpose of these layers is to reliably transfer the data from the sending device to the receiving device. The layers on top of the transport layer—the session, presentation, and application layers—as you will soon see, are concerned more with application issues and making sure data is formatted the right way. Two types of protocols are defined at this layer: connection-oriented and connectionless protocols.

A connection-oriented protocol establishes an end-to-end connection between the sender and the receiver, keeps track of all the segments that are being transmitted, and has a retransmission mechanism in place. Retransmissions happen in cases in which some of the segments that are sent are lost on the way and don't get to their destination or become corrupted and cannot be recovered at the receiver. A system of acknowledgments is in place to signal successfully receiving the data and availability to receive additional data. A sliding window mechanism dynamically adjusts the amount of data that is transmitted on the wire before the sender is waiting for an acknowledgment from the receiver. Large data transfers, SSH terminal sessions, and World Wide Web and email sessions use connection-oriented protocols at the transport layer. Although not defined with the OSI specification, Transmission Control Protocol (TCP) is the most common connection-oriented protocol at the transport layer.

A connectionless protocol does not establish an end-to-end connection between the sender and the receiver. This type of data transmission is mostly used for real-time traffic such as voice and video, in which it is better to ignore the data segments that are lost or corrupted on the way than to stop the transfer of data and ask for retransmission. For example, in the case of a voice over IP (VoIP) call, the end-user experience is much better if the network just ignores a few milliseconds of lost data that the human ear can compensate for anyway than if the network stops the flow of data, queues the data already received at the receiver, retransmits the missing few milliseconds, replays the message, and then dequeues the data that was waiting for playback at the receiver. User Datagram Protocol (UDP) is the most common connectionless protocol used at the transport layer.

The **session layer** establishes, manages, and terminates sessions between two communicating devices. On top of the end-to-end communication channel that the transport layer establishes, the session layer has the role of keeping track of whose turn is to transmit data, and it ensures that the same operation is not performed at the same time by different clients or even by resuming failed transmissions. Applications that implement the Remote Procedure Call (RPC) framework use the session layer extensively.

The **presentation layer** ensures that the information from the application layer of the sending device is readable and interpretable by the application layer of the receiving device. The presentation layer is not focused on the moving of bits between devices as much as it is focused on how the data being transferred is organized and the syntax and semantics used for this data. For example, if different encodings for text-based exchange of information are used on the sending and receiving devices (for example, ASCII versus EBCDIC), the presentation layer uses a common encoding format for translating between the two encodings and makes sure data at the application layer is displayed correctly on both systems. Also at this layer, the data is differentiated and encoded based on the type of information it contains, such as text files, video files, binary files, and so on.

The **application layer** is the OSI layer that is closest to the user and contains all the applications and protocols that the users are interacting with, such as email, file transfer, and video streaming. One application layer protocol that is extensively used is Hypertext Transfer Protocol (HTTP), which is at the heart of the World Wide Web. Whenever a website is accessed on a web server, the data between the client browser and the web server is carried over HTTP.

The OSI model didn't find much success when it was initially developed, and it was not adopted by most network equipment vendors at the time, but it is still used for education purposes and to emphasize the importance of separating data transmission in computer networks into layers with specific functions.

The TCP/IP Model

A much more successful implementation of the layered architecture to data transmission and device networking is the TCP/IP (Transmission Control Protocol/Internet Protocol) reference model, also referred to as the *Internet Protocol suite*. Created around the same time as the OSI model, the TCP/IP reference model was adopted and implemented by most networking vendors back in those days and has over time become the reference model for the Internet.

The Internet Protocol suite and the main protocols that make up the model, TCP and IP, started as a research project by the U.S. Department of Defense through a program called Defense Advanced Research Projects Agency (DARPA) in the 1960s. It was the middle of the Cold War, and the main purpose of the project was to build a telecommunications network that would be able to withstand a nuclear attack and still be able to function if any of the devices making up the network were destroyed or disabled.

Much like the OSI model, the TCP/IP reference model uses a layered approach (see Figure 16-2). Each layer provides the functionality and features specified in the reference model and serves the layer above it.

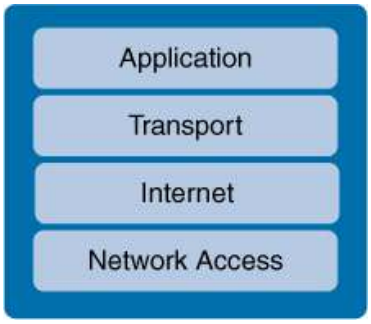


Figure 16-2 *TCP/IP Reference Model Layers*

The network access layer in the TCP/IP model corresponds to the physical and data link layers in the OSI model, and it provides the same services to the layer above. It deals with moving data packets between the Internet layer interfaces of two devices connected on the same link. It specifies the physical characteristics of serial lines, Ethernet, and wireless links, such as voltage levels, data transmission rates, maximum transmission distance, and physical connectors. The network layer defines how data is formatted for transmission on the physical link, how the data packet is received from the Internet layer for transmission by adding a header and footer, how the data frames are transmitted over the physical medium, how access to the network is controlled, and which device can transmit at which point. Switching and switches, as described later in this chapter, operate at the network access layer.

The Internet layer of the TCP/IP reference model corresponds to the network layer of the OSI model in terms of functions and characteristics. It is responsible for transmitting data across disparate and distant networks in a process called *data routing*. *Routers* are devices that operate at the Internet layer and perform the routing function. As you might have guessed from the name of the reference model, Internet Protocol (IP) is the main protocol operating at the Internet layer. The IP protocol performs two basic functions:

- Device addressing and identification
- Data packet routing

Devices connected to a TCP/IP network are identified by their IP addresses. Two addressing solutions are currently supported on the Internet: IP version 4 (IPv4) and IP version 6 (IPv6). IPv4 uses addressing based on 32 bits, which means that approximately

4 billion devices can be uniquely identified on the network. As you can imagine, there are more than 4 billion endpoints connected to the Internet today. Temporary solutions to address the lack of IPv4 addresses have been developed over the years, including private IP address subnets and Network Address Translation (NAT), both of which are discussed later in this chapter. The permanent fix to this problem was the development and standardization in 1998 of IPv6, which uses 128-bit addresses and is able to provide unique addresses for a gigantic number of endpoints and devices. Both IPv4- and IPv6-addressed devices are currently supported and connected to the Internet, but IPv4 is slowly being phased out.

The second function of the Internet layer is data packet routing. Routing packets means forwarding data from its source toward the destination through a network of interconnected routers. Through this functionality, the Internet layer makes possible internetworking and the connection of different IP networks, essentially establishing the Internet. The packet forwarding at the Internet layer is best effort and unreliable, and any error correction and retransmission mechanisms required have to be implemented at higher layers. The Internet layer is agnostic to the data structures and the operations that the layer above it is implementing. If we go back to the postal mail analogy, the Internet layer acts as the postal service infrastructure and the mail carriers who deliver the letters and packages without knowing what they actually contain. In the same way, the Internet layer receives data segments from the transport layer and carries them all the way to the destination without knowing what the actual data is.

The transport layer is at the core of the TCP/IP reference model. It establishes end-to-end data channels over which applications exchange data, regardless of the structure of the user data and the underlying network. Since the Internet layer is best effort, the protocols at the transport layer provide flow control, congestion control, segmentation, and error control. Differentiating between different applications running on the same host occurs at the transport layer through port numbers. The transport layer port is a 16-bit logical construct allocated for each of the communications channels an application requires. The ports in the range 0 to 1023, called *well-known ports* or *system ports*, are used for system processes that provide widely used types of network services. For example, web servers that implement HTTP are by default listening for connections on port 80 for regular HTTP traffic or on port 443 for secure HTTP traffic via HTTP Secure (HTTPS). End-to-end connections at the transport layer can be categorized as either connection oriented (usually implemented using TCP) or connectionless (using UDP).

TCP uses several different technologies to ensure reliable bit stream delivery. First of all, data is delivered in the order in which it was sent, and TCP has mechanisms of keeping track of each data segment sent. A dynamic windowing system controls traffic congestion and ensures that the sender doesn't overwhelm the receiver with too much data. If packets are lost on the way to the destination or become corrupted in transit, TCP handles the retransmission of this data. If the data becomes duplicated in transmission, the duplicate data is discarded at the receiver.

UDP is a connectionless datagram protocol that is used in situations where timely data delivery is more important than reliability. It is a best-effort, unreliable protocol that implements a basic checksum mechanism for error detection. UDP is typically used for applications such as streaming media, voice, and video.

The TCP/IP model transport layer corresponds roughly to the OSI model transport layer described earlier in this chapter.

The application layer, as the name suggests, contains all the end-user applications that need to exchange data over the network. The application layer protocols exchange data on top of the end-to-end, sender-to-receiver connection that is established by the lower-layer protocols. There are a large number of application layer protocols, including the following:

- Hypertext Transfer Protocol (HTTP) is used for transferring web pages between web browsers and web servers.
- File Transfer Protocol (FTP) is used for transferring files between a client and a server.
- Dynamic Host Configuration Protocol (DHCP) is used to dynamically assign IP addresses to devices on a network.

The application layer in the TCP/IP reference model corresponds to the session, presentation, and application layers of the OSI model.

In order to be able to communicate between devices using a layered model, the protocols running at each layer on the source host must be able to communicate with their counterparts at the destination host. The data that is being exchanged at each layer is called a protocol data unit (PDU). Based on the layer processing the data, the PDUs have different names:

- At the application layer, the general term for the PDU is *data*.
- At the transport layer, the PDU is called a *segment*.
- At the Internet layer, the PDU is called a *packet*.
- At the data link layer, the PDU is called a *frame*.

Figure 16-3 shows the PDUs at each layer and the flow of data from sender to receiver.

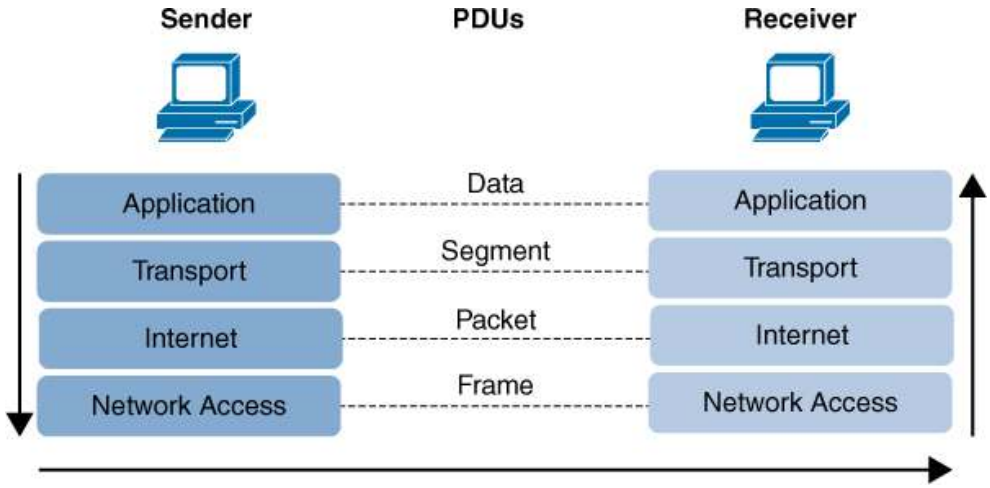


Figure 16-3 *TCP/IP Reference Model PDUs at Each Layer*

As data moves down the layers at the sending host, it gets processed at each layer. This process is called *encapsulation*, and there is a corresponding process at the destination host called *de-encapsulation*. The encapsulation process consists in adding extra protocol information at each layer so that layer-to-layer communication can take place. Starting at the top, the user data generated in the application layer gets passed down to the transport layer for transmission. The protocol running at the transport layer takes the data it received from the application layer, adds its header information, and passes the datagram PDU to the Internet layer. Similarly, the Internet layer adds its own header and passes the packet PDU to the data link layer. The data link layer takes the data from the Internet layer and builds a frame PDU that is ready to be sent on the physical medium by adding its own protocol header and a frame check sequence (FCS) footer. The role of the FCS footer is to ensure that the data is transmitted without errors. At each hop along the way to the destination, a checksum of the whole frame is computed and added to the FCS footer. If the computed checksum is different from the checksum in the FCS footer, the receiving device knows an error occurred on the transmission path. At the data link layer, the user information with all the header and footer information is finally transmitted on the physical medium.

As the data is received at the destination host, it goes through a similar process but in the reverse order. This process, called *de-encapsulation*, consists of removing the header and footer information at each layer and passing the data to the layer above until it gets to the application layer, where it is interpreted by the application protocol. Figure 16-4 shows the processes of encapsulation at the sender and de-encapsulation at the receiver.

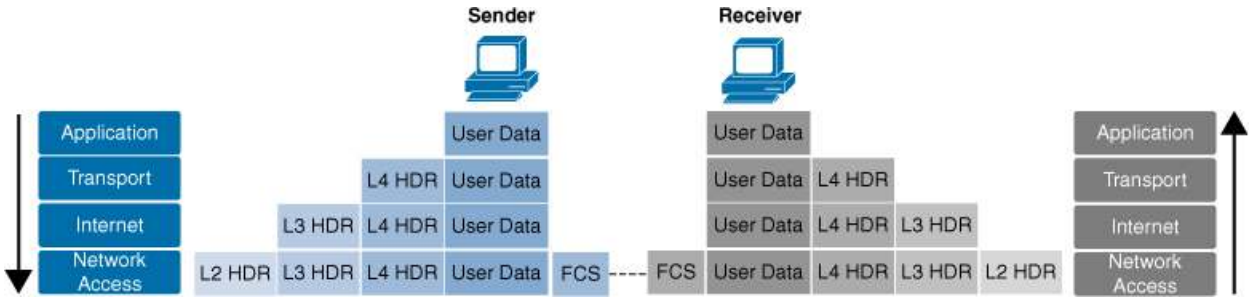


Figure 16-4 *Encapsulation and De-encapsulation*

Switching Concepts

Data frame switching is a critical process in moving data traffic from a source network endpoint to a destination endpoint within a local-area network (LAN). This section introduces several networking concepts: Ethernet, MAC addresses, VLANs, and switching.

Ethernet

A LAN is a network that is confined within a building or campus. These types of networks start on the smaller side with home networks, where end-user devices such as personal computers, smartphones, laptops, and printers connect—usually through a wireless connection— to a local home network. The main characteristic of these networks is that the data traffic between the devices connected to the LAN stays local and can be transferred between these devices by a *switch*, which is a network device that operates at the data link layer and enables direct communication between devices connected to a LAN.

The most common LAN technology is Ethernet. Ethernet is a data link layer protocol defined by the IEEE. The Ethernet protocol encompasses guidelines and standards that specify cabling and signaling formats at the physical and data link layers. For example, Ethernet standards specify characteristics for different types of cables and network interface ports, and they also specify voltage levels and the maximum distance the signal can travel before it becomes too attenuated to be successfully received. Over

time, the Ethernet protocol has evolved from transfer rates of 10 Mbps to 100 Gbps and even 400 Gbps. The transmission medium has also evolved from coaxial cable to twisted-pair cable, fiber optics, and wireless.

As mentioned previously, the PDU at the data link layer is called a *frame*. The Ethernet protocol specifies the format shown in Figure 16-5 for the Ethernet frame, which includes the following fields:

Field Length (Bytes)	8	6	6	2	46-1500	4
Typical Ethernet Frame	Preamble	Destination Address	Source Address	Type	Data	FCS

Figure 16-5 *Ethernet Frame Format*

- **Preamble:** This field consists of 8 bytes of alternating 1s and 0s that are used to synchronize the signals of the sender and receiver.
- **Destination Address:** This field contains the address of the receiver.
- **Source Address:** This field contains the address of the source device.
- **Type:** This field contains a code that identifies the network layer protocol.
- **Data:** This field contains the data that was received from the network layer and that needs to be transmitted to the receiver.
- **Frame Checksum Sequence (FCS):** This 4-byte field includes a checksum mechanism to ensure that the frame has been transmitted without corruption.

MAC Addresses

Media Access Control (MAC) addresses are used to enable communication between devices connected to a local network. Several types of MAC addresses are used to accommodate the different types of network communications. There are three major types of network communications:

- **Unicast:** In this type of communication, data frames are sent between one specific source and addressed to one specific destination. This type of transmission has one sender and one receiver, and it is the most common type of traffic on any network.
- **Broadcast:** In this type of communication, data frames are sent from one source address to all other addresses connected on the same LAN. There is one sender, but the information is sent to all devices connected to the network.
- **Multicast:** In this type of communication, information is sent from one device to a group of devices or clients. In order for the clients to receive the multicast data, they have to be members of a multicast group. Whereas broadcast is used to transmit data to all the clients on the network, multicast is used to transmit data to just a subset of those clients.

In order to distinguish between these three types of network communications, the IEEE has defined MAC addresses for each type:

- When the least significant bit of a MAC address's first byte is 0, it means the frame is meant to reach only one receiving NIC, which means unicast traffic.
- When the least significant bit of the first octet of a MAC address is 1, it means the frame is a multicast frame, and the receiving NICs will process it if they were configured to accept multicast MAC addresses. An example of a multicast MAC address that is used by Cisco Discovery Protocol (CDP) is 01-00-0C-CC-CC-CC.
- When all the bits are set to 1, it means the frame is a broadcast frame, and it is being received by all the NICs on that network segment.

Each client that needs to exchange data on an Ethernet network needs to have a unique MAC address so that the data is directed to the proper destination device. The MAC address is burned into the network interface card (NIC) and is also called the burned-in address (BIA) or the physical address of the device. A MAC address has 48 bits organized as 12 hexadecimal numbers. There are several different ways of representing MAC addresses, all of them containing the same information. The following representations of a MAC address are all equivalent:

- 0000.0c59.beef
- 00:00:0c:59:be:ef
- 00-00-0C-59-BE-EF

A MAC address has two components (see Figure 16-6):

- **Organizationally unique identifier (OUI):** This 24-bit number identifies the manufacturer of the NIC. The IEEE assigns OUIs to NIC manufacturers.
- **Vendor-assigned address:** This 24-bit number uniquely identifies the Ethernet hardware.

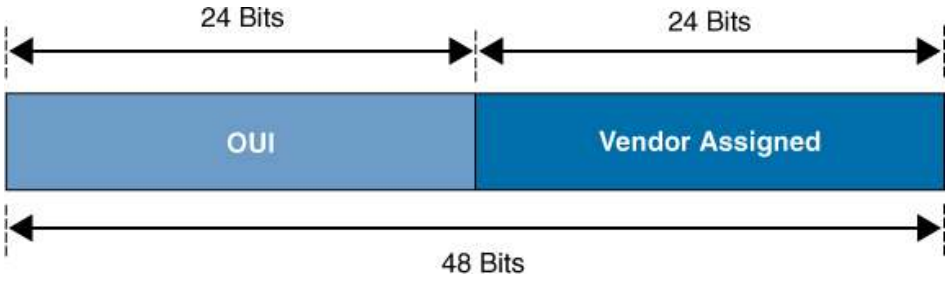


Figure 16-6 *MAC Address Components*

Virtual Local-Area Networks (VLANs)

A virtual local-area network (VLAN) is a group of devices on an Ethernet network that are logically segmented by function, team, or application. Imagine a university campus network and all the devices that need access to it. It is a common practice to combine into the same logical construct devices that need the same type of access to the network or that perform the same function. For example, the student devices will probably be grouped into the student VLAN, the faculty devices into the faculty VLAN, the finance department devices into the finance VLAN, and so on. On top of the physical network that connects these devices, a software layer is added to separate them into different silos. This is done following the IEEE 802.1q standard, which specifies an additional field in the data link layer frame: the 802.1q header. This header is 4 bytes long and is wedged between the source MAC address field and the Type field, as shown in Figure 16-7.

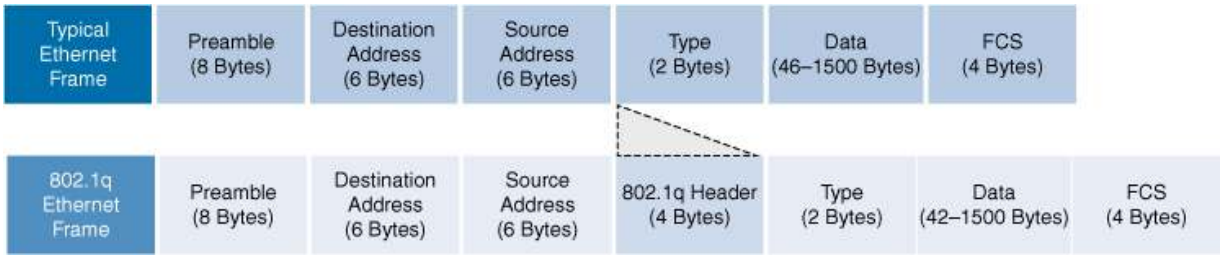


Figure 16-7 *Typical Ethernet Frame Versus 802.1q Ethernet Frame*

In the 802.1q header, 12 bits are dedicated to the VLAN identifier, so a maximum of 4094 VLANs can be defined on an Ethernet network; the VLANs with IDs 0 and 4095 are reserved. Each VLAN defines its own broadcast domain. A data link layer broadcast domain is defined as the subset of devices on a network that receive Layer 2 broadcast traffic. Layer 2 broadcast traffic stops and is not forwarded further by Layer 3 devices or routers. It is a best practice to keep the Layer 2 broadcast domain as small as possible. Going back to the university campus example, imagine the thousands of devices connected to the network as being part of the same broadcast domain. In addition, because there is no Time to Live (TTL) field in the Ethernet frames, and redundant connections are required between switches, broadcast storms might bring down the network and make it unusable. Creating VLANs and assigning the devices to specific VLANs limits the broadcast domain, reduces the chance of a broadcast storm, and logically groups the devices for easier troubleshooting. For each VLAN, a dedicated Layer 3 IP subnet is usually allocated. It is much easier to enforce security policies and limit access between VLANs at a Layer 3 device than it is to accomplish the same on a port-by-port basis on a switch. No matter the physical location in the network, devices can be part of the same VLAN (see Figure 16-8).

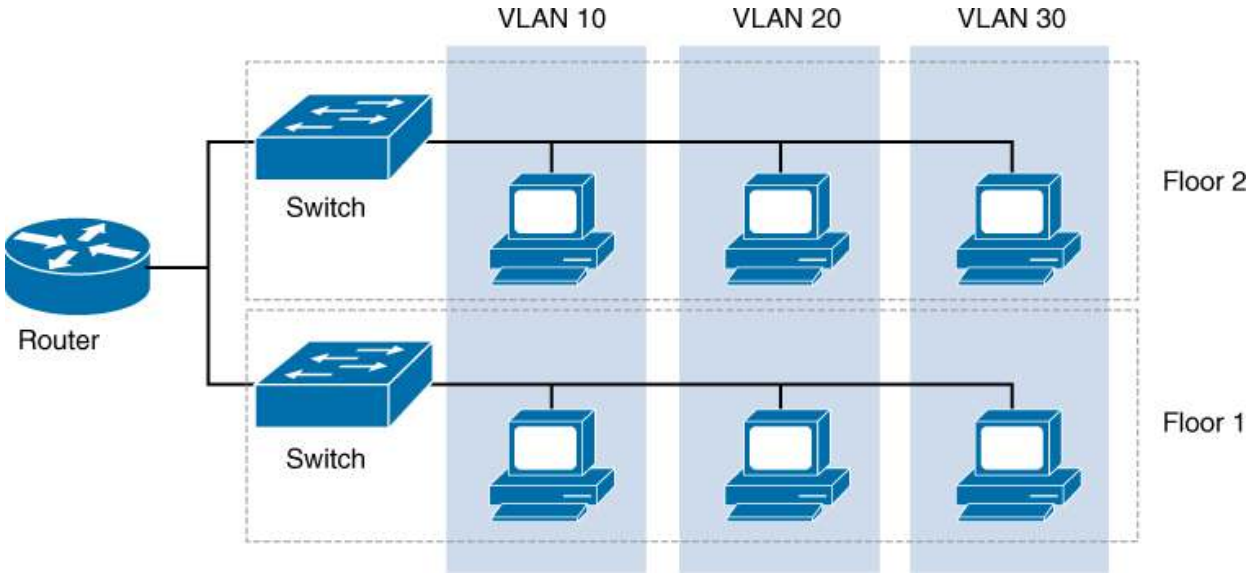


Figure 16-8 *Virtual Local-Area Networks (VLANs)*

VLANs provide network segmentation by reducing the broadcast domains and organizational flexibility by combining devices on a network, based on the needs of the organization.

Switching

Switching is the process through which a data frame is forwarded from its source toward its destination by a Layer 2 device called a *switch*. In a typical LAN, all devices connect to the network either through an Ethernet port available on the NIC or through a wireless NIC that connects to a wireless access point that is connected to a switch port. In the end, whether wired or wireless, all client devices connect to Ethernet switches. Forwarding of data traffic between devices at Layer 2 is based on the MAC address table that is stored on each switch within the network. The MAC address table is dynamically built and contains a list of all the MAC addresses that the switch has learned and the switch ports on which it learned those addresses.

A MAC address table on a switch might look as shown in Table 16-2.

Table 16-2 *MAC Address Table*

VLAN	MAC Address	Type	Ports
10	001b.10a0.2500	Dynamic	Gi0/1
20	001b.10ae.7d00	Dynamic	Gi0/2
30	0050.7966.6705	Dynamic	Gi0/3

Based on this table, the switch can forward the Layer 2 data frames toward their destination. If a data frame with MAC destination address 0050.7966.6705 arrives at the switch with this MAC address table, the switch will forward that frame out its GigabitEthernet0/3 port toward the host with that specific MAC address.

The switch dynamically builds and maintains the MAC address table and indicates where each device is connected in the network. Let's assume that a switch was just rebooted, and its MAC address table is empty. As the devices that are connected to the switch start exchanging data, the MAC address table gets populated with their addresses. From the first device that sends data on the network, the switch looks at the source MAC address in the data frame and records it in the MAC address table, together with the port on which it received the data. Since this is the first MAC address in the table, the switch doesn't have the destination MAC address of the frame, so it floods it over all the ports except the port on which it received the data. As the frame gets flooded throughout the network, eventually it will get to its destination. The destination device replies with its own data frame, in which the source and destination MAC addresses get swapped. All the other devices drop the original frame because it was not destined for them. The switch receives the frame, takes note of the source MAC address, and records it in its table, together with the port on which it was received. This way, the switch dynamically builds its MAC address switching table and can forward traffic between all the devices that are connected to it.

Routing Concepts

Routing, or Layer 3 packet forwarding, is the process of selecting a path through a network. To understand routing, you need to understand IPv4 and IPv6 addresses. Then you can learn about the routing process itself.

IPv4 Addresses

The most common protocol at the Internet layer is Internet Protocol (IP). The Internet, the worldwide network connecting billions of devices and users, is built on top of IP. There are currently two versions of IP available: IP version 4 (IPv4) and IP version 6 (IPv6). Both protocol versions are used in the Internet, and a slow migration toward IPv6 is in progress.

The main role of IP is to uniquely identify devices at the Internet layer in the TCP/IP reference model and Layer 3 in the OSI model. IPv4 uses 32-bit addresses, which means it can theoretically support 2^{32} (just over 4 billion) unique addresses. It is a connectionless protocol, IP data packets are forwarded individually, the delivery of packets is best effort, and there is no retransmission or data recovery functionality specified in the protocol definition. In case of lost or corrupt packets, retransmission and data recovery are left to be implemented by the higher-layer protocols. Since the physical medium and transmission are handled by the data link layer, all those specifications are abstracted from the Internet layer, and IP operates independently of the medium that is carrying the data.

IP addresses are similar to physical addresses of homes and businesses. The same way the address of a house uniquely identifies it so that packages and other mail can be delivered, the IP address uniquely identifies all the devices connected to the Internet so that data packages can be exchanged and delivered. For easier readability for humans, IPv4 addresses are split into four octets, with each decimal value separated by a dot—in a format known as dotted decimal notation. Each octet is made up of 8 bits, and each bit can have a value of 1 or 0. For example, an IP address would take the following form in the IP header: 1100000010101 0000011000001000001. It is very easy to make mistakes when representing IP addresses this way. The four-octet dotted decimal notation for the same address becomes 192.168.48.65, which is much easier to remember and keep track of. Table 16-3 shows decimal and binary representations of this IPv4 address.

Table 16-3 *IPv4 Address Representation with Decimal and Binary Values*

Decimal	192	168	48	65
Octet/binary	11000000	10101000	00110000	01000001

An IP address consists of two parts:

- **Network ID:** The network address part starts from the leftmost bit and extends to the right. Devices on a network can communicate directly only with devices that are in the same network. If the destination IP address is on a different network than the network the source IP address is on, a router needs to forward the traffic between the two networks. The router maintains a routing table with routes to all the networks it knows about.
- **Host ID:** The host address part starts from the rightmost bit and extends to the left. The host ID uniquely identifies a specific device connected to the network. Although the host ID can be the same between different devices on different networks, the combination of network ID and host ID must be unique throughout the network.

In order to accommodate networks of different sizes, IP addresses are organized into different classes. There are in total five classes of IPv4 addresses. Classes A, B, and C are available for public use; Class D is used for multicast addresses; and Class E is reserved for research and is not available for public use. The Internet Assigned Numbers Authority (IANA) manages the assignment and allocation of IP addresses to Internet service providers (ISPs), which in turn assign IP address ranges to their customers. The five classes of IPv4 addresses are described in Table 16-4.

Table 16-4 *Classes of IPv4 Addresses*

Address Class	Bit Pattern of First Byte	First-Byte Decimal Range	Host Assignment Range in Dotted Decimal Format
A	0xxxxxxx	1 to 127	1.0.0.1 to 127.255.255.254
B	10xxxxxx	128 to 191	128.0.0.1 to 191.255.255.254
C	110xxxxx	192 to 223	192.0.0.1 to 223.255.255.254
D	1110xxxx	224 to 239	224.0.0.1 to 239.255.255.254
E	11110xxx	240 to 255	240.0.0.1 to 255.255.255.255

Class A uses the first byte for the network ID and the other 3 bytes (or 24 bits) for the host ID. As you can imagine, networks with 2^{24} (more than 16 million) hosts in the same network are nonexistent; technologies such as classless interdomain routing (CIDR) and variable-length subnet masking (VLSM) have been developed to address the wastage of IPv4 addresses with the original definition of classes of IPv4 addresses. The first bit of the first byte in a Class A IP address is always 0. This means that the lowest number that can be represented in the first byte is 00000000, or decimal 0, and the highest number is 01111111, or decimal 127. The 0 and 127 Class A network addresses are reserved and cannot be used as routable network addresses. Any IPv4 address that has a value between 1 and 126 in the first byte is a Class A address.

Class B uses the first 2 bytes for the network ID and the last 2 bytes for the host ID. The first 2 bits of the first byte in a Class B IPv4 address are always 10. The lowest number that can be represented in the first byte is 10000000, or decimal 128, and the highest number that can be represented is 10111111, or decimal 191. Any IPv4 address that has a value between 128 and 191 in the first byte is a Class B address.

Class C uses the first 3 bytes for the network ID and the last byte for the host ID. Each Class C network can contain up to 254 hosts. A Class C network always begins with 110 in the first byte, meaning it can represent networks from 11000000, or decimal 192, to 11011111, or decimal 223. If an IP address contains a number in the range 192 to 223 in the first byte, it is a Class C address.

The role of the address mask is to identify which portion of the IP address is the network ID part and which portion is the host ID. Similarly to IPv4 addresses, address masks are also 32 bits long and use dotted decimal notation. IPv4 addresses and address masks go together hand in hand. Address masks specify the network ID portion of an IP address by turning the corresponding bit into a 1 and the host ID portion of the IP address by turning the corresponding address mask bit into a 0. The default address masks for the first three classes of IPv4 addresses are as follows:

- **Class A:** The default address mask is 255.0.0.0 or /8, indicating that the first 8 bits of the address contain the network ID, and the other 24 bits contain the host ID.
- **Class B:** The default address mask is 255.255.0.0 or /16, indicating that the first 16 bits of the address contain the network ID, and the last 16 bits contain the host ID.
- **Class C:** The default address mask is 255.255.255.0 or /24, indicating that the first 24 bits of the address contain the network ID, and the last 8 bits contain the host ID.

In each IP network, there are two reserved addresses. The IP address where all the host ID bits are 0 is reserved and is used to identify the network itself. For example, the Class C network address 192.16.1.0 cannot be assigned to any host on that network because it is used to identify the network. This is how the network is represented in the routing tables of routers throughout the network. The other reserved IP address is the one where all the host ID bits are 1; this address, called the *broadcast address*, is used to send information to all the hosts on the network and cannot be assigned to any device. In the previous example with the Class C network, the broadcast address for that network would be 192.16.1.255. Class C networks can accommodate and uniquely address 256 − 2 = 254 devices on each network.

The Internet has experienced explosive growth, and it won't be long before IANA runs out of available IPv4 address ranges to assign. A large number of the IPv4 address ranges that were initially allocated were also used in private networks, so IANA decided to create a group of private IPv4 address ranges for those networks. Private IPv4 networks, defined by the Internet Engineering Task Force (IETF) in RFC 1918: *Address Allocation for Private Internets*, are meant to preserve the IPv4 address space by having dedicated networks for private networks. Three blocks of IP addresses (1 Class A, 16 Class B, and 256 Class C networks) are reserved for private, internal use (see Table 16-5).

Table 16-5 *Private IPv4 Addresses*

Address Class	Private Network ID	Network Address Range
A	10.0.0.0	10.0.0.0 to 10.255.255.255
B	172.16.0.0	172.16.0.0 to 172.31.255.255
C	192.168.0.0	192.168.0.0 to 192.168.255.255

IP addresses in these ranges are not routable in the Internet. When private networks using these IPv4 addresses need to connect to the Internet, their addresses need to be translated to public, Internet-routable IPv4 addresses. This translation process is called Network Address Translation (NAT), and it is discussed in more detail in Chapter 18, "IP Services."

The need to further divide IPv4 address classes, especially the Class A and Class B networks, and to create additional subnetworks became clear as networks started to grow in size and complexity and IPv4 addresses started to run out. As previously mentioned, a Class A network with the default address mask can address more than 16 million devices on the same network. Having more than 16 million devices on one Layer 3 network is theoretically possible but practically impossible. Class B networks have 16 bits dedicated to host IDs, which means each Class B network can address 2^{16} = 65,536 − 2 = 65,534 devices (with two addresses reserved for the network address and the broadcast address). These networks also are in need of further subnetting. Subnetting is the process of borrowing bits from the host ID portion of a network and transforming them into network ID bits.

Classless interdomain routing (CIDR) goes beyond the class limits and the default address masks and represents a way of specifying the subnet mask for an IP address. Initially when the classes of IPv4 addresses were defined, the default masks for each class were already implied and were not even mentioned when assigning networks for customer consumption. As the available IPv4 addresses became more and more scarce, especially for Class A and Class B networks, CIDR was introduced as a way to capture the transition from a class-based Internet to a classless one. In a classless Internet, it is no longer feasible to assume that the address mask for a Class A network is 255.0.0.0 or /8 or for a Class B network that is 255.255.0.0 or /16. The purpose of the / notation is to specify how many bits in the subnet mask are dedicated to the network ID. Network addresses can be subdivided further to create *subnets*. This is accomplished by using variable-length subnet masking (VLSM) to indicate how many bits in an IPv4 address are dedicated to the network ID and how many bits are dedicated to the host ID. As an example, the 10.0.0.0 private network can be further subdivided, and new subnets can be created out of it. Instead of having one network with more than 16 million hosts, which is not helpful in real life, 65,536 subnets with 254 hosts—each similar to 10.0.0.0/24, 10.0.1.0/24, 10.0.2.0/24, and so on—can be created out of the 10.0.0.0/8 network. This is done by borrowing 16 bits from the 24 bits that are dedicated to host IDs and transforming them into network ID bits. By applying a different subnet mask—using /24 instead of /8—we can use VLSM and CIDR notation to adjust the host portion of the network address.

Let's take a practical example and create new subnets for a fictitious enterprise called Acme, Inc. The network administrator is planning on using the 192.168.0.0/24 network to address all the devices that are connected to the Acme, Inc. enterprise network. As you would expect, there are several departments in the Acme, Inc. world, and there are different numbers of employees in each department. Let's assume that the total number of departments is 8 and include the typical corporate organization: engineering, support, human resources, finance and accounting, and so on. Acme, Inc. is still a small company, and there are not more than 20 devices connected to the network in each of the 8 departments. The network administrator would like to segment each department into its own network so that security can be more easily enforced by using Layer 3 access control lists (ACLs) and also to limit the broadcast domain for each department. The 192.168.0.0/24 CIDR notation indicates that for this subnet, there are 24 bits reserved for the network ID, and the rest of the bits, 8, are reserved for the hosts. In order to create 8 subnets out of the original 1, 3 bits ($2^3 = 8$) would have to be borrowed from the host portion and dedicated to the new network IDs. The subnet mask in this case changes as follows:

- **Original subnet mask:** 11111111.11111111.11111111.00000000 (/24)
- **New subnet mask:** 11111111.11111111.11111111.11100000 (/27)

There are still 5 bits left for hosts on each subnet, which results in $2^5 = 32 - 2$ (1 reserved for the network address and 1 reserved for the broadcast address) = 30 usable IP addresses for each subnet. This is more than needed to accommodate the 20 devices for each department. The new subnets look like those shown in Table 16-6; each one has a /27 mask:

Table 16-6 *Creating Additional Subnets from 192.168.0.0/24*

Subnet	Network Address	Broadcast Address	Available Host Address Range
Subnet 1	192.168.0.0	192.168.0.31	192.168.0.1 to 192.168.0.30
Subnet 2	192.168.0.32	192.168.0.63	192.168.0.33 to 192.168.0.62
Subnet 3	192.168.0.64	192.168.0.95	192.168.0.65 to 192.168.0.94
Subnet 4	192.168.0.96	192.168.0.127	192.168.0.97 to 192.168.0.126
Subnet 5	192.168.0.128	192.168.0.159	192.168.0.129 to 192.168.0.158
Subnet 6	192.168.0.160	192.168.0.191	192.168.0.161 to 192.168.0.190
Subnet 7	192.168.0.192	192.168.0.223	192.168.0.193 to 192.168.0.222
Subnet 8	192.168.0.224	192.168.0.255	192.168.0.225 to 192.168.0.254

New subnets can be easily created to accommodate any number of devices. In the previous example, each subnet can address up to 30 devices, but the subnets can be further subnetted, if needed. By borrowing additional bits from the host section, subnet masks of /28, /29, and even /30 can be specified to create new subnets for $2^4 = 16 - 2 = 14$, $2^3 = 8 - 2 = 6$, or $2^2 = 4 - 2 = 2$ hosts in each subnet. /30 subnets are especially useful on point-to-point serial connections between two routers—for example, where there are only 2 devices connected and there is no point in using larger subnets as that would only lead to wasted addresses. Table 16-7 provides a breakdown of the subnet masks from /24 to /32, their binary representations, and the number of subnets and hosts for each subnet.

Table 16-7 *Subnet Mask Representations and the Number of Subnets and Hosts for Each*

Decimal Subnet Mask	Binary Subnet Mask	CIDR	Subnets	Hosts per Subnet
255.255.255.255	11111111.11111111.11111111.11111111	/32	None	N/A
255.255.255.254	11111111.11111111.11111111.11111110	/31	None	N/A
255.255.255.252	11111111.11111111.11111111.11111100	/30	64	2
255.255.255.248	11111111.11111111.11111111.11111000	/29	32	6
255.255.255.240	11111111.11111111.11111111.11110000	/28	16	14
255.255.255.224	11111111.11111111.11111111.11100000	/27	8	30
255.255.255.192	11111111.11111111.11111111.11000000	/26	4	62
255.255.255.128	11111111.11111111.11111111.10000000	/25	2	126
255.255.255.0	11111111.11111111.11111111.00000000	/24	1	254

IPv6 Addresses

IPv4 has valiantly served the Internet from the early days, when only a handful of devices were interconnected at college campuses in the United States for research purposes, to today, when billions of devices are interconnected and exchanging massive amounts of data. The longevity of IPv4 and the addressing scheme it defines is a testament to how robust and scalable IP actually is. Even so, IPv4 faces limitations, including the insufficiency of 32-bit addresses, the need for the IP header to be streamlined and simplified, and the need for security embedded in the protocol rather than implemented as an afterthought.

IPv6 is an evolutionary upgrade to IP that is meant to address most of the shortcomings of IPv4. IPv6 is defined in RFC 2460, *Internet Protocol, Version 6 (IPv6) Specification*, issued by the IETF. Several other RFCs that have been created following the initial definition describe the architecture and services supported by IPv6. IPv6 addresses are 128 bits long instead of 32 bits. The 128 bits provide 3.4×10^{38} addresses; this is the equivalent of a full IPv4 addressing space for each living person on Earth. Currently both IPv4 and IPv6 addresses are supported on the Internet, and a gradual transition to only IPv6 is happening, but we won't be there until sometime far in the future. Several migration technologies have been developed to ease the transition from IPv4 to IPv6. The dual-stack method involves having both IPv4 and IPv6 addresses configured on the same interface and seems to be the most popular option. Tunneling mechanisms are also available for IPv4 tunneling over IPv6 networks and IPv6 tunneling over IPv4 networks.

IPv6 brings several improvements compared to IPv4, including the following:

- Larger address space means improved global reachability and better aggregation of IP prefixes, which leads to smaller routing tables, which leads to faster routing.
- End-to-end communication is now possible, as there is no need for technologies such as Network Address Translation (NAT) or Port Address Translation (PAT) that have been developed to address the lack of public IPv4 addresses.
- Address autoconfiguration allows hosts and devices connecting to the network to easily and automatically obtain IPv6 addresses. This is especially important for mobile devices as they roam between different networks; autoconfiguration makes the transition seamless.
- The coexistence of multiple addresses on the same interface increases reliability and load balancing.
- A simplified header means better routing efficiency and increased forwarding performance.
- No broadcast addresses and no broadcast traffic mean no broadcast storms, improving the reliability of the whole network.
- Security is increased, thanks to built-in IPsec capabilities.

IPv6 addresses can be displayed as 32 hexadecimal characters split into 8 sections, as follows:

- Colons separate entries *x*:*x*:*x*:*x*:*x*:*x*:*x*:*x*, where *x* is a 16-bit hexadecimal field.
- Hexadecimal characters are case insensitive.
- Leading zeros in a field are optional.
- Successive fields of zero can be represented as :: once per address.

An example of an IPv6 address is 2031:0000:120F:0000:0000:09C0:9A73:242C. Following the guidelines just specified, this address can also be written as follows:

2031:0000:120F::9C0:9A73:242C (because successive fields of zero can be represented as ::)

2031:0:120F::9C0:9A73:242C (because leading zeros in a field are optional)

Figure 16-9 shows a simple illustration of how IPv6 addresses and subnetting work.

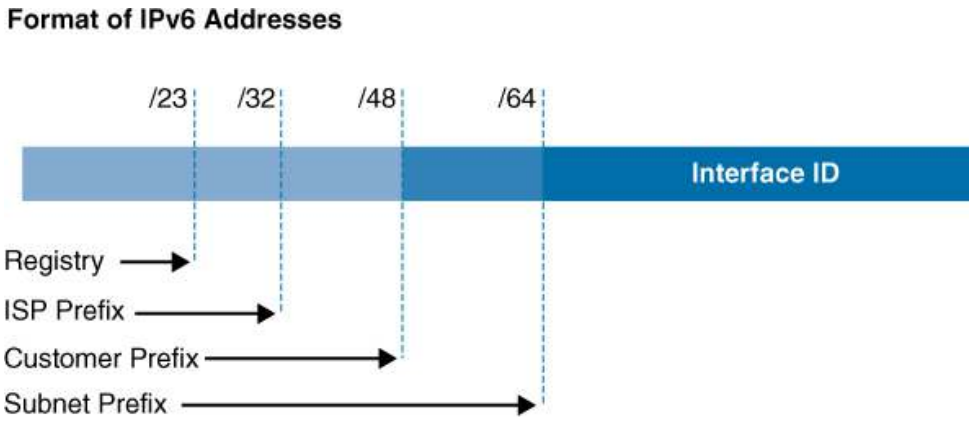


Figure 16-9 *IPv6 Subnetting*

The Internet Assigned Numbers Authority (IANA) is the organization that manages the assignment of IPv6 addresses at a global level. IANA assigns /23 blocks of IPv6 addresses to the registries that manage IPv6 addresses at a regional level, such as the Asia-Pacific Network Information Center (APNIC) or the American Registry for Internet Numbers (ARIN). The regional registries assign /32 blocks of IPv6 addresses to the Internet service providers (ISPs) that operate in their geographic areas. The ISPs, in turn, assign /48 IPv6 subnets to their customers, which can subnet further and create their own subnets, usually down to a /64, giving them up to 65,536 IPv6 subnets that they can assign to their sites.

The interface ID portion of an IPv6 address is usually based on the MAC address of the device, following the EUI-64 specification. This specification creates the 64 bits needed for the interface ID portion of the address by splitting the 48-bit MAC address right down the middle and inserting the 16-bit 0xFFFE between the OUI and the vendor-assigned bits.

The hierarchical distribution of IPv6 addresses allows for better aggregation of routes, which leads to smaller Internet routing tables and faster routing.

There are several different types of IPv6 addresses:

- Global
- Link-local
- Multicast
- Loopback
- Unspecified

Global IPv6 addresses are the equivalent of public unicast IPv4 addresses. Global IPv6 addresses are assigned in a controlled fashion so that they can be easily aggregated at all levels, registry, ISPs, and customers. Currently IANA is assigning IPv6 addresses that start with the binary value 001 (2000::/3). This represents one-eighth of the total IPv6 address space. A global unicast IPv6 address typically consists of a 48-bit global routing prefix in the most significant portion of the address, followed by a 16-bit subnet ID and 64 bits dedicated to the interface identifier. Addresses with prefixes in the range 2000::/3 to E000::/3 are required to have 64-bit interface identifiers that follow the EUI-64 format mentioned previously.

Link-local addresses in IPv6 are a new concept compared to IPv4. These addresses refer only to a particular physical link and are not forwarded by routers. They are used only for local communication between devices on the same physical network segment. Link-local addresses are part of the link-local prefix FE80::/10 and are extensively used in IPv6 routing protocols for neighbor and router discovery, as well as in automatic address configuration. They are usually dynamically created on all IPv6 interfaces by adding the 64-bit interface ID to the FE80::/10 prefix.

IPv6 multicast addresses are part of the FF00::/8 prefix. There is no broadcast traffic in IPv6, so that functionality is taken by multicast and anycast traffic in IPv6. There are several reasons broadcast traffic has been left out of IPv6, including the fact that each broadcast packet on the network generates an interrupt on all the devices that are part of the same broadcast domain, even if the traffic is not directly meant for them; in addition, broadcast storms can bring down entire networks. With IPv6, it was decided to implement the one-to-many type of data traffic by using multicast. Multicast increases the efficiency of delivering data from one device to many, and with IPv6, it gets a larger address space, too. A data packet sent to a multicast address is delivered to all interfaces identified by the multicast address. Multicast addresses also have scope. The scope of a multicast address can be that of

a node, a link, a site, an organization, or a global scope and is represented by a scope parameter of 1, 2, 5, 8, or E, respectively. As an example, the multicast address with the prefix FF02::/16 is a multicast address with a link scope. All devices implementing IPv6 are required to join the all-nodes multicast group with the FF02:0:0:0:0:0:0:1 multicast address. IPv6 routers must also join the all-routers multicast group FF02:0:0:0:0:0:0:2. The scope on both of these multicast addresses is link-local.

Loopback IPv6 addresses are similar to the loopback addresses in IPv4: They are used for testing local traffic on a device. In IPv6 there is only one address dedicated for this purpose: 0:0:0:0:0:0:0:1 (or ::1 when using the zero compression feature in IPv6).

The unspecified address is used for situations when a device doesn't have any IPv6 address assigned and needs to populate the Source field when sending a datagram on the network, such as when obtaining an IPv6 address from a DHCPv6 server. This address is the all-zeros address: 0:0:0:0:0:0:0:0 (or just :: when using the zero compression feature in IPv6).

Anycast addresses are allocated from the unicast global space. Their purpose is to accommodate the one-to-nearest data traffic requirement. In this case, multiple devices share the same unicast address, and data traffic that is destined for the anycast address is routed through the network to the closest device that has that anycast address configured on one of its interfaces. In this case, of course, all the anycast nodes have to provide a similar, uniform service. Anycast is suitable for load balancing and content delivery services. An example of a service that can be implemented using anycast addresses is Domain Name System (DNS). DNS is covered in Chapter 18; for the purposes of this example, you just need to know that DNS is a critical component of the Internet and is mainly used to resolve domain names to IP addresses. Configuring several DNS servers on a network with the same anycast address ensures a consistent DNS configuration on all the devices in the network. Instead of having to configure several different DNS server IP addresses for redundancy purposes, in IPv6, a single address, the anycast address, can be configured on all DNS clients. As long as at least one of the DNS servers that have been configured as part of the anycast group is online, DNS redundancy is guaranteed.

Routing

Routing is the process of selecting a path through a network. This functionality is performed by routers, which are devices made to interconnect networks, find the best paths for data packets from their source toward their destination, and route data based on a routing table. As switches have a critical role at Layer 2 in connecting devices in the same physical area and creating local-area networks, routers have a critical role at Layer 3 in interconnecting networks and creating wide-area networks. The main functions of a router are the following:

- Path determination
- Data packet routing or forwarding

Path determination is the process through which routers look up information in their routing tables to determine where to forward the data packets received on their interfaces. In a typical scenario, a router has at least two interfaces—and usually more. Each interface connects to a Layer 3 network, which means the interface on the router is configured with an IP address in that network. In most cases, the router acts as the default gateway for that Layer 3 network, meaning that any traffic that is not destined for the local network will be forwarded to the router, and the router will forward it further toward its destination. Each router maintains its own routing table, which contains a list of all the destinations or networks that are known to the router and how to reach those destinations. When receiving a packet on one of its interfaces, the router checks the destination IP address in the data packet header and looks up the best match for that destination in its routing table. If the destination IP address matches one of the networks in the routing table, it means that either the network is directly connected to the router or it can be reached via another router that is directly connected, and it becomes the next-hop router toward the final destination of the data packet. If there is no match for that destination IP address in the routing table, the router sends the data packet to the default route, and if there is no default route, the router drops the data packet.

After determining the correct path for the packet, the router forwards the packet through a network interface toward the destination. Figure 16-10 shows a simple network diagram.

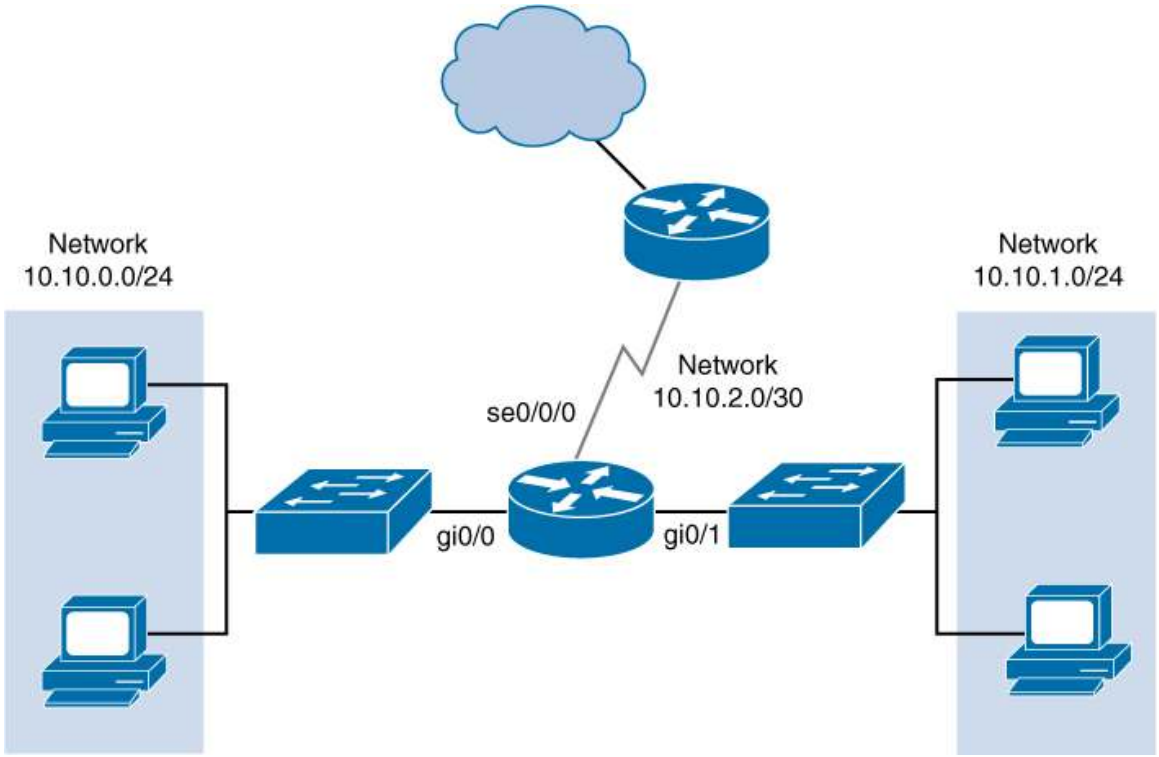


Figure 16-10 *Simple Network Diagram*

The routing table for the network in Figure 16-10 might look like Table 16-8.

Table 16-8 *Routing Table*

Network	Interface or Next Hop
10.10.0.0/24	directly connected: Gi0/0
10.10.1.0/24	directly connected: Gi0/1
10.10.2.0/30	directly connected: Se0/0/0
0.0.0.0/0	via 10.10.2.2 (next-hop router)

Each row in the routing table lists a destination network and the corresponding interface or next-hop address. For directly connected networks, the router has an interface that is part of that network. For example, let's assume that a router receives a data packet on its GigabitEthernet0/1 interface with destination IP address 10.10.0.15. The router looks up the destination address in its routing table and decides to route the packet out its GigabitEthernet0/0 interface toward the destination.

Following the same logic, let's assume next that the router receives a packet on its GigabitEthernet0/0 interface with destination IP address 172.16.0.25. The router performs a routing table lookup and finds that it doesn't have an explicit route for that network, but it does have a default route (0.0.0.0/0) via the next-hop router with IP address 10.10.2.2. Looking up recursively how to reach the 10.10.2.0 network, the router forwards the packet out its Serial0/0/0 interface toward the next-hop router, which should be closer to the destination. Default routes on routers—much like the default gateway configurations on end host devices—are used as a last resort for routing any data packets for which the destination doesn't explicitly exist in the routing table. Default routes are extremely helpful in maintaining a small routing table and hence decreasing the routing table lookups and also decreasing the amount of time needed to decide on which interface to route the traffic, leading to faster routing of data packets. For the majority of routers, a full Internet routing table with hundreds of thousands of routes is not necessary, and a default route toward the next hop is sufficient.

There are several ways in which routing tables are populated:

- Directly connected networks
- Static routes
- Dynamic routes
- Default routes

We've already discussed directly connected networks, in which the router has an interface configured for that network and is routing traffic for it and the default routes, which are used as a last resort in the event that a more specific route doesn't exist in the routing table.

Static routes are configured manually by the network administrator or the person managing a router. In small networks in which there are not expected to be any changes, static routes can work just fine. In larger networks that are more dynamic in nature, with devices and routers going offline and coming online constantly, static routes do not work very well because the administrator needs to manually change the entries as the changes happen in the network. A much more scalable solution in this case is dynamic routes.

Dynamic routes, as the name implies, are learned dynamically through specialized software applications called *dynamic routing protocols*. Several routing protocols have been developed over the years, including Open Shortest Path First (OSPF), Enhanced Interior Gateway Routing Protocol (EIGRP), Routing Information Protocol (RIP), Border Control Protocol (BGP), and Intermediate System-to-Intermediate System (IS-IS). Each of these protocols has its own characteristics, advantages, and drawbacks. It is the role of the network administrator to decide which protocol to choose to run on the network and to determine whether a dynamic routing protocol is even necessary in the first place. Medium and large networks run at least one of these protocols on all the routers in the network. The routers running dynamic routing protocols exchange data between them and dynamically build their routing tables based on the information they receive from their neighbor routers. If multiple routing protocols are configured on the routers, there needs to be a way to decide which routes to choose and to populate the routing table with them. Each routing protocol has an administrative distance associated with it that is based on the trustworthiness of the protocol. For example, if a route for the same network is learned through RIP and OSPF, the OSPF route will take precedence as OSPF has a lower administrative distance than RIP. A lower administrative distance means a higher degree of trust in that source —so a higher chance of getting added to the routing table.

Connected routes, meaning the router has an interface configured in a specific network, have the lowest administrative distance (AD), 0. Static routes are preferred next as they usually have an AD of 1. Routing protocols have different administrative distance values; for example, BGP has an AD of 20, and RIP has an AD of 120. These values for AD are default values, and they can be changed if additional tweaking of the routing table is necessary. All dynamic routing protocols mentioned support routing for both IPv4 and IPv6.

As we've seen previously in this chapter, the Internet has become classless, and there are a large number of networks and subnets that can be created from those networks. The same way that a large network address space can be subdivided into small subnets, the reverse process is also available; it is called *supernetting*. This means that smaller subnets can be combined into a larger network, and instead of having several routing table entries for each individual subnet, a single routing table entry can encompass them all. For example, when an ISP assigns a certain prefix range to one of its customers, the ISP does not need to know all the details of how the customer is implementing subnetting in its internal network. All the ISP needs to know is that the whole prefix is assigned to the customer, and it can be reached via the customer's border routers. Supernetting is critical in maintaining an Internet routing table of a manageable size.

Another important aspect when building a routing table is the longest prefix match concept. Longest prefix match means that when the router is looking in the routing table for a match to route the data packet toward its destination, it does so by trying to match it to the longest prefix it has in the table. For example, let's assume that the routing table has two entries: 10.10.10.0/24 via interface GigabitEthernet0/0 and 10.10.0.0/16 via interface GigabitEthernet0/1. If the router receives a data packet destined for 10.10.10.15, it looks in its routing table, and it will match the longest-prefix network (in this case, 10.10.10.0/24) and route the packet out its GigabitEthernet0/0 interface. If the data packet has destination address 10.10.20.15, the router matches the other entry, 10.10.0.0/16, since that is the longest prefix match it has for that subnet.

Exam Preparation Tasks

As mentioned in the section "How to Use This Book" in the Introduction, you have a couple of choices for exam preparation: the exercises here, Chapter 19, "Final Preparation," and the exam simulation questions on the companion website.

Review All Key Topics

Review the most important topics in this chapter, noted with the Key Topic icon in the outer margin of the page. Table 16-9 lists these key topics and the page number on which each is found.

Table 16-9 *Key Topics*

Key Topic Element	Description	Page Number
Paragraph	OSI model's layered approach	485
Paragraph	TCP/IP model's layered approach	488
Paragraph	Communication between devices in a layered model	490
Paragraph	Client MAC addresses	493
Paragraph	VLAN identifier	494
Paragraph	MAC address table for a switch	495
Paragraph	IP in the TCP/IP reference model and the OSI model	496
List	Improvements of IPv6 over IPv4	501
List	Types of IPv6 addresses	502
List	Ways routing tables are populated	506

Define Key Terms

Define the following key terms from this chapter and check your answers in the glossary:

Open Systems Interconnection (OSI)
Transmission Control Protocol/Internet Protocol (TCP/IP)
Media Access Control (MAC)
Logical Link Control (LLC)
User Datagram Protocol (UDP)
Transmission Control Protocol (TCP)
Internet Protocol (IP)
Hypertext Transfer Protocol (HTTP)
File Transfer Protocol (FTP)
protocol data unit (PDU)
frame check sequence (FCS)
organizationally unique identifier (OUI)
virtual local-area network (VLAN)
Internet Assigned Numbers Authority (IANA)
classless interdomain routing (CIDR)
variable-length subnet masking (VLSM)
Asia-Pacific Network Information Center (APNIC)
American Registry for Internet Numbers (ARIN)

Chapter 17. Networking Components

This chapter covers the following topics:

- **What Are Networks?:** This section describes a network and introduces the types of networks.
- **Elements of Networks:** This section overviews various networking elements, including hubs, switches, routers, and VLANs.
- **Software-Defined Networking:** This section overviews the fundamentals of the control, data, and management planes.

Over the past several decades, Cisco has built the backbone of the internet today and has developed and created big and small networks, including enterprise and service provider networks. In Chapter 16, "Network Fundamentals," you learned some networking basics. This chapter covers networking components, such as the following:

- Networking elements such as hubs, switches, and routers
- Network device functions, including the management, data, and control planes
- Software-defined networking

"Do I Know This Already?" Quiz

The "Do I Know This Already?" quiz allows you to assess whether you should read this entire chapter thoroughly or jump to the "Exam Preparation Tasks" section. If you are in doubt about your answers to these questions or your own assessment of your knowledge of the topics, read the entire chapter. Table 17-1 lists the major headings in this chapter and their corresponding "Do I Know This Already?" quiz questions. You can find the answers in Appendix A, "Answers to the 'Do I Know This Already?' Quiz Questions."

Table 17-1 *"Do I Know This Already?" Section-to-Question Mapping*

Foundation Topics Section	Questions
What Are Networks?	1–3
Elements of Networks	4–7
Software-Defined Networking	8–9

Caution

The goal of self-assessment is to gauge your mastery of the topics in this chapter. If you do not know the answer to a question or are only partially sure of the answer, you should mark that question as wrong for purposes of self-assessment. Giving yourself credit for an answer that you correctly guess skews your self-assessment results and might provide you with a false sense of security.

1. Modern switched Ethernet LANs most commonly use which physical topology?

   a. star

   b. ring

   c. bus

   d. mesh

2. _____-area networks usually use public networks like telephone, cellular, or satellite networks.

a. Metropolitan

b. Local

c. Wide

d. Campus

3. Which of the following is the least secure in terms of privacy of the user?

a. Router

b. Hub

c. Switch

d. Bridge

4. Which of the following is not true of a Layer 3 switch?

a. It uses IP addresses for forwarding.

b. A VLAN can be implemented by using this type of switch.

c. A Layer 3 switch creates a single broadcast domain.

d. A Layer 3 switch usually supports Layer 2 switching as well.

5. Which of the following can be used to segment a physical switch into multiple logical networks?

a. Bridge

b. WAN

c. VLAN

d. Router

6. FIB tables, which contain precomputed routes, reverse lookups, and so on, are used in which type of switching?

a. Process switching

b. Cisco Express Forwarding

c. Fast switching

d. None of the above

7. What is NAT used for?

a. Stopping local traffic from reaching the public network

b. Mapping private IP addresses to public IP addresses

c. Detecting traffic patterns

d. Assigning IP addresses to local network clients

8. In a network controller, which plane is concerned with administrative access to a network device?

a. Control

b. Management

c. Data

d. Hardware

9. In a Cisco SD-WAN solution, what is the main brain of the entire solution that manages the control plane?

a. Cisco SD-WAN Manager

b. vSmart

c. vEdge

d. vBond

Foundation Topics

What Are Networks?

NIST CNSSI 4009 defines a network as follows:

information system(s) implemented with a collection of interconnected components. Such components may include routers, hubs, cabling, telecommunications controllers, key distribution centers, and technical control devices.

In other words, a network is formed when two or more systems are connected and can communicate with each other. With different types of connections, various types of networks can be built. Three different factors are used to classify networks:

- Topology
- Standard
- Size or location

Topology refers to how the various nodes or elements connect. Figure 17-1 shows three prominent network topologies:

- **Bus:** In a bus network, all the elements are connected one after the other.
- **Star:** In a star topology, all nodes in the system are connected to a central point.
- **Ring:** In a ring topology, each node connects to two neighboring nodes to form a closed loop. Token passing is an access method used by Token Ring and some other ring protocols, not a defining property of every ring topology.

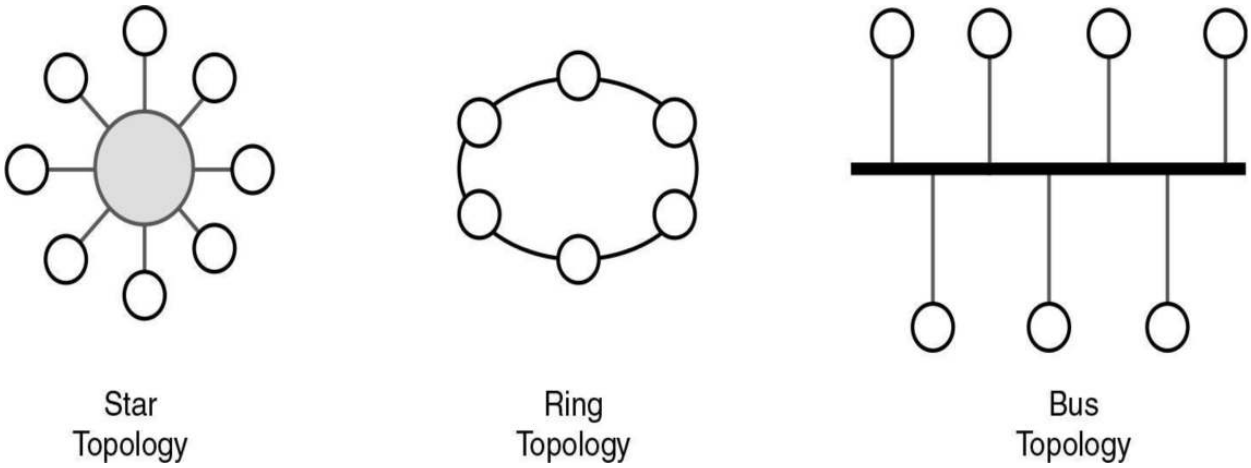


Figure 17-1 *Network Topologies*

The standard refers to the protocol that the network uses. Ethernet is one example of a standard. Modern switched Ethernet commonly uses a physical star topology, although Ethernet supports multiple physical arrangements. Ethernet data rates range from 10 Mbps through multigigabit and much higher-speed variants. Another standard is Token Ring, which, as the name suggests, is a ring topology type. Token Ring rates vary from 4 Mbps to 16 Mbps.

The size of a network is a bit of an arbitrary concept. Network sizes fall into basically four network types:

- **Local-area network (LAN):** A LAN is typically a single network for an office, an enterprise, or a home, and it is located entirely within a single facility. Figure 17-2 shows an example of a LAN comprising various laptops, PCs, servers, and printers.

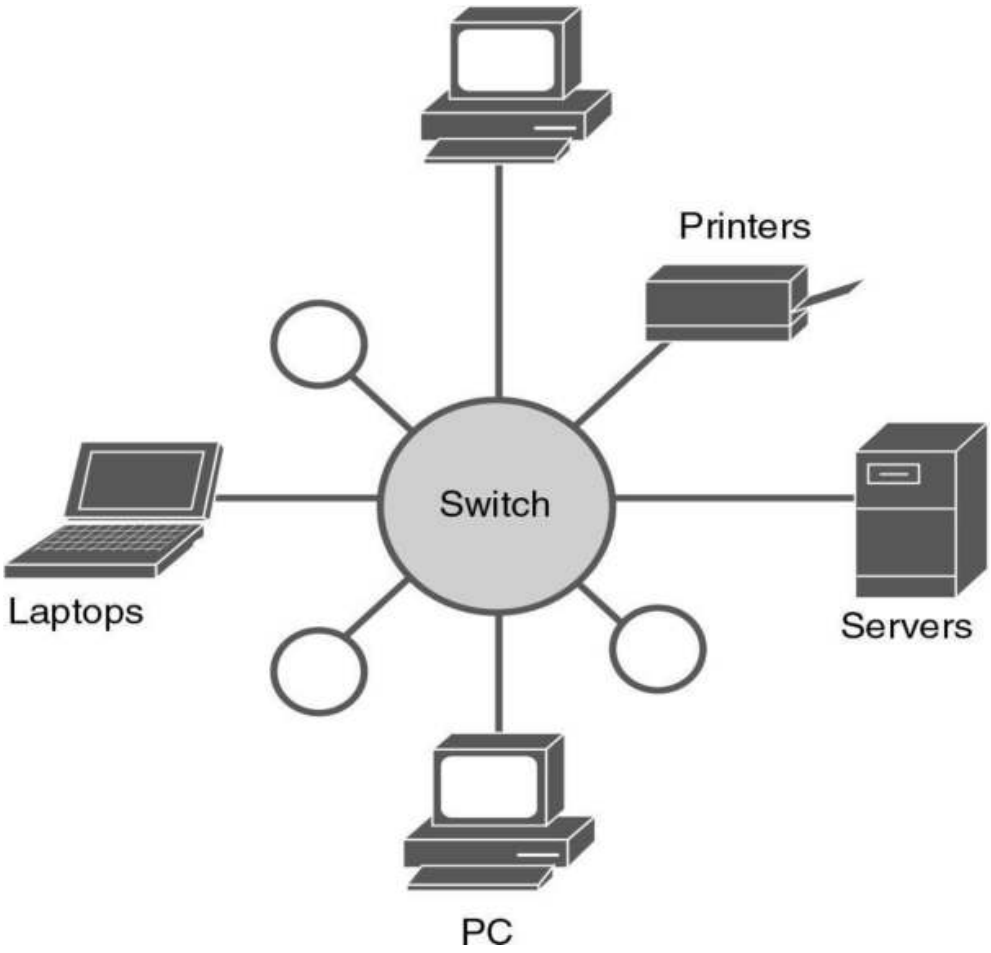


Figure 17-2 *Local-Area Network*

- **Campus-area network (CAN):** A CAN consists of two or more LANs within a limited area. CANs help interconnect various buildings or departments within a campus, an enterprise, a large office, or a school. Figure 17-3 shows an example of a CAN that connects various LAN segments.

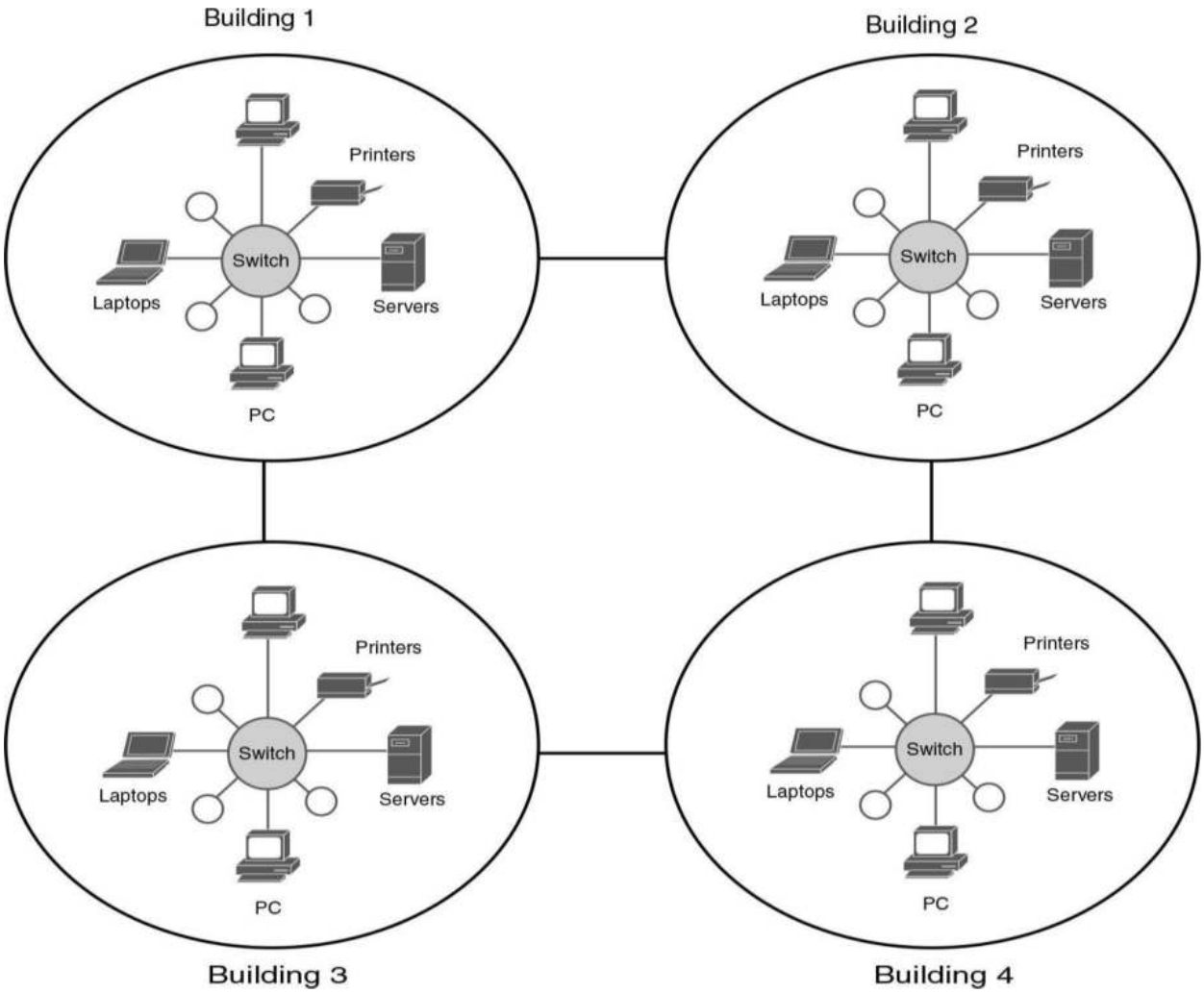


Figure 17-3 *Campus-Area Network*

- **Metropolitan-area network (MAN):** A MAN is a network that is extended to cover multiple locations within a slightly larger distance than a CAN. A classic example of a MAN is the cable network in a city. Figure 17-4 shows an example of a MAN that connects various network segments.

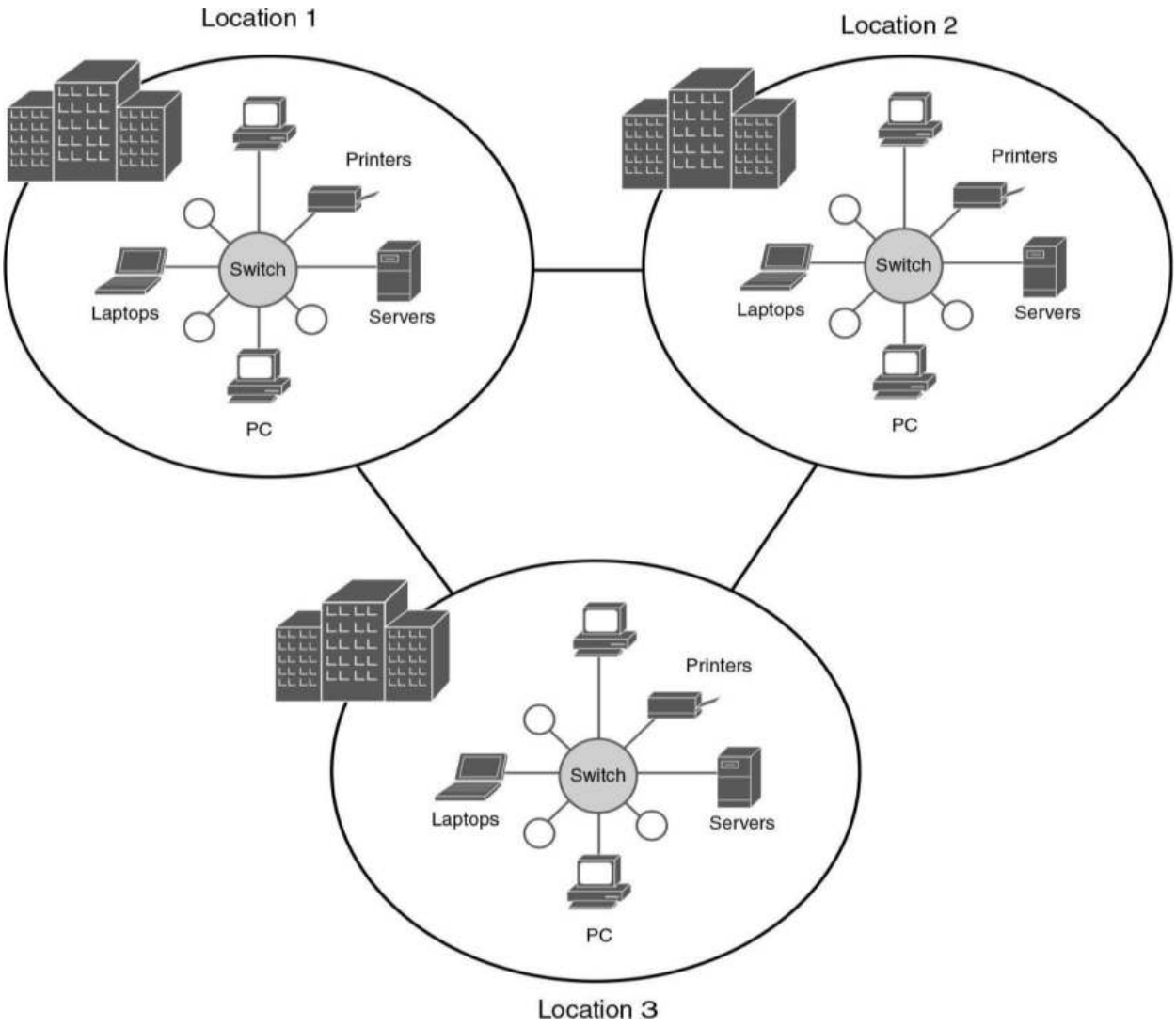


Figure 17-4 *Metropolitan-Area Network*

- **Wide-area network (WAN):** A WAN covers an even larger geographic area than a MAN. The connections are through public networks, such as the telephone system, microwaves, satellite links, or leased lines. Figure 17-5 shows an example of a WAN that connects various remote offices with the headquarters.

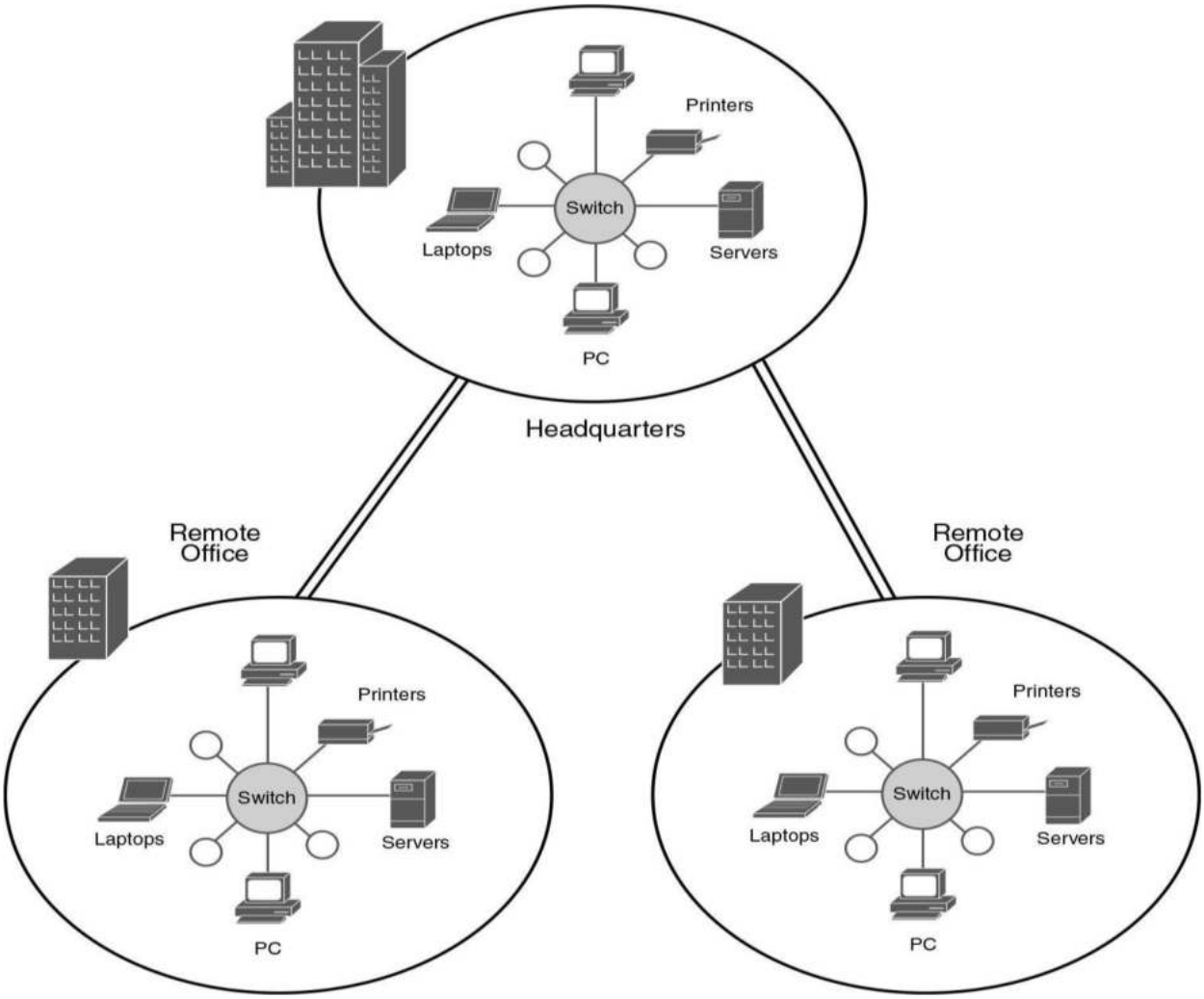


Figure 17-5 *Wide-Area Network*

Elements of Networks

Now that you have seen the various types of networks, let's look at what constitutes a network. A network has several basic components:

- Hubs
- Switches
- Bridges
- Routers

The following sections look at each of these elements in detail and the differences between them. In the following sections, you will see how to build a simple home network in which you want to connect two or three computers or devices.

Hubs

A hub is the least intelligent and most rudimentary network device. With a hub, traffic or data packets in any port are always sent out to all other ports. For example, in Figure 17-6, Laptop 1 wants to communicate with Device 3. The hub replicates and sends the packet from the laptop to each port. Only Device 3 responds, and that response is again replicated to all ports. Every device connected to the hub "sees" everything. It is up to the devices to decide whether a packet is for them and whether they should respond. The hub remains passive and does not know or understand the traffic that flows through it.

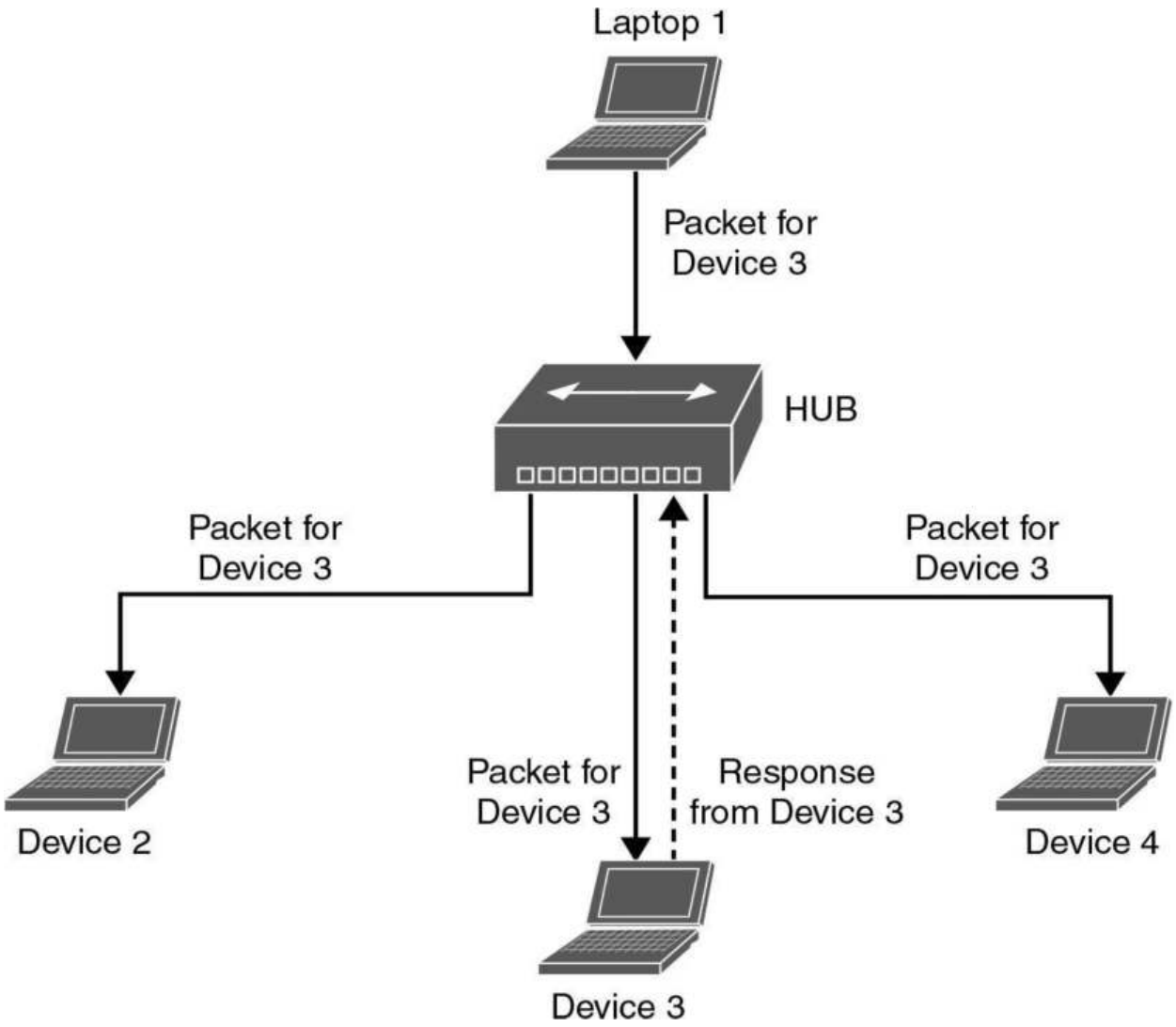


Figure 17-6 *Hub: Every Device Receives Every Packet*

One of the biggest challenges with hubs is that every frame shows up at every device attached to a hub instead of just showing up at the intended destination. This can be a huge security problem. Another challenge has to do with performance: Because a hub can receive a lot of inbound frames, sending out frames can be difficult and can result in performance issues on the network.

Bridges

A bridge connects two physical network segments that use the same protocol. Each network bridge builds a MAC address table of all connected devices on its ports. When packet traffic arrives at the bridge, whose target address is local to that side, the bridge filters that frame so it stays on the local side.

If the bridge cannot find the target address on the side that received the traffic, it forwards the frame across the bridge, hoping the destination will be on the other network segment. In some cases, multiple bridges are cascaded.

Figure 17-7 shows how a bridge connects two LAN segments.

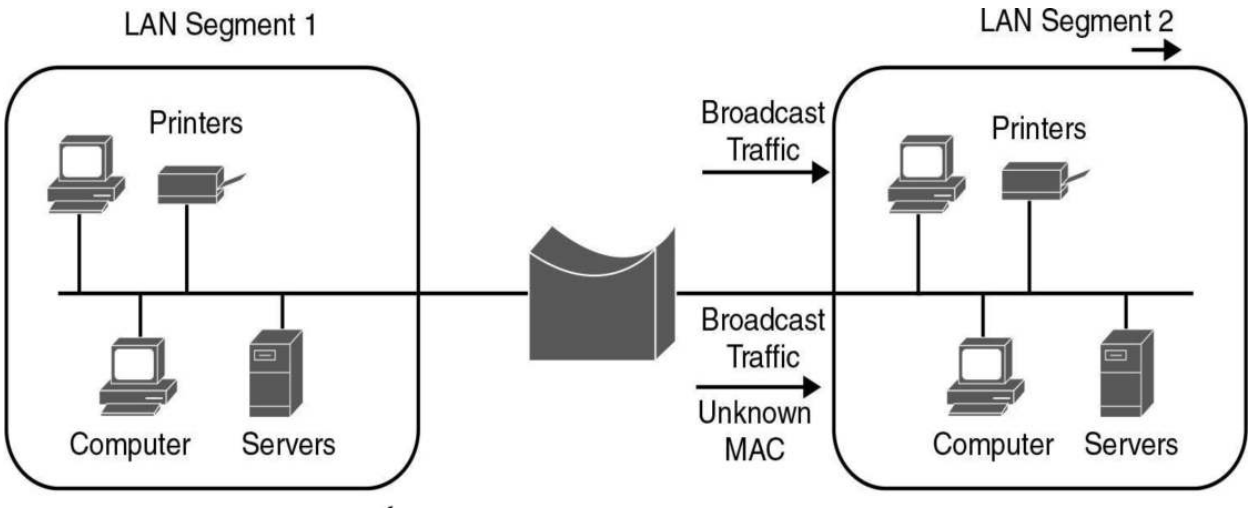
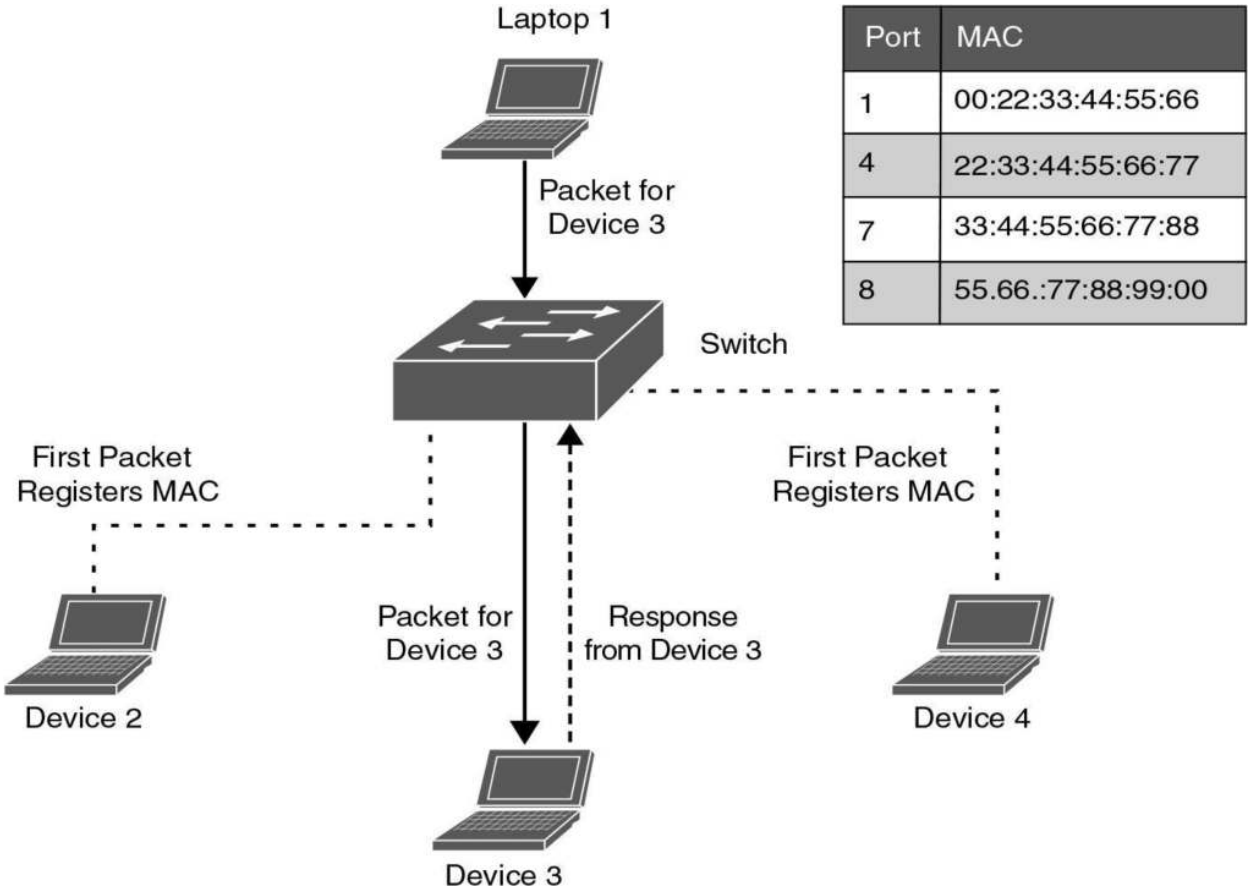


Figure 17-7 *Using a Bridge to Connect Two LAN Segments*

Switches

You can think of a switch as an "intelligent" hub. A switch does what a hub does—but more efficiently and intelligently. It analyzes the traffic as it flows. A switch builds a port-to-MAC address table as devices start communicating. This means a switch offers dramatically better performance than a hub. Specialized hardware (called application-specific integrated circuits [ASICs]) is needed to analyze the traffic. If we replace the hub with a switch in our example, we get the setup shown in Figure 17-8, where a Layer 2 switch maintains a port-to-MAC address lookup table.



Port	MAC
1	00:22:33:44:55:66
4	22:33:44:55:66:77
7	33:44:55:66:77:88
8	55.66.:77:88:99:00

Figure 17-8 *Using a Layer 2 Switch for Port-to-MAC Address Mapping*

As you can see in Figure 17-8, each of the devices is "seen," and the MAC address is registered when the first packet is sent by the device. Over a period, the switch generates a table that contains the port number on which each MAC address was seen. It

then uses this table to determine the destination MAC address and sends the incoming packet to the correct outgoing port. In the figure, a packet destined for Device 3 is sent only to the proper port.

Recall our look at the OSI model layers in Chapter 16. The switch described in this section is a Layer 2 switch, which works at Layer 2 of the OSI model (the data link layer). The switch sends a packet to a destination port using the MAC address lookup table, which stores the device's MAC address associated with each port. In contrast, a Layer 3 switch works at the network layer of the OSI model (Layer 3). It performs a destination-prefix lookup in a routing table or forwarding information base (FIB), then uses adjacency information for the Layer 2 rewrite and outbound interface. Figure 17-9 shows this Layer 3 forwarding process.

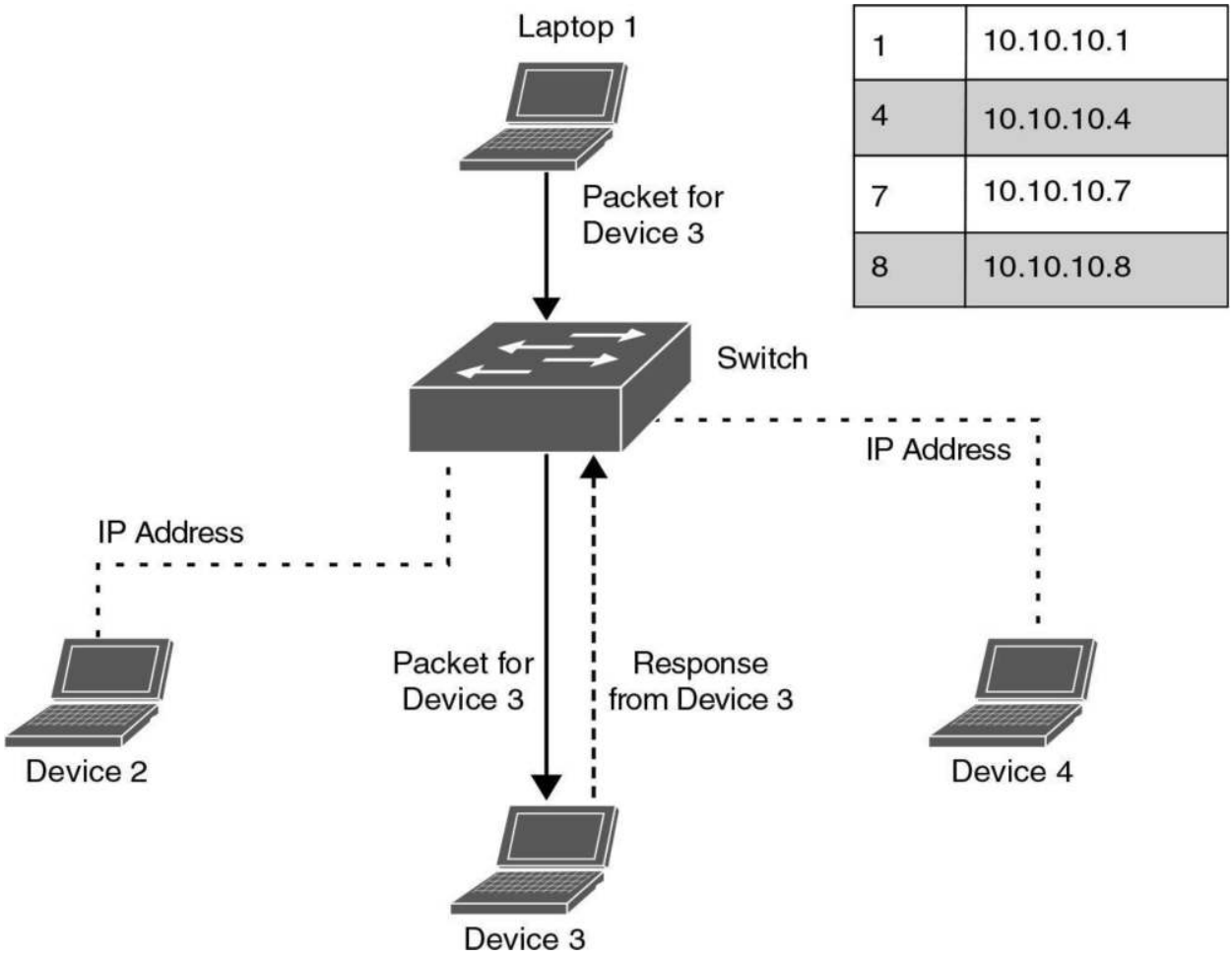


Figure 17-9 *Using a Layer 3 Switch for IP Forwarding*

Table 17-2 looks at some differences between Layer 2 and Layer 3 switches.

Table 17-2 *Differences Between Layer 2 and Layer 3 Switches*

Layer 2 Switch	Layer 3 Switch
Operates at Layer 2 (data link) of the OSI model	Operates at Layer 3 (network layer) of the OSI model
Uses MAC addresses for forwarding	Uses IP addresses for forwarding
Performs Layer 2 switching only	Performs both Layer 2 and Layer 3 switching
Can reduce unnecessary local traffic through segmentation	Can route between VLANs and IP subnets
Supports one or more broadcast domains through VLANs	Routes between multiple broadcast domains

Virtual Local Area Networks (VLANs)

A switch can be logically segmented into multiple broadcast domains using virtual LANs (VLANs). If you have one switch, you can create multiple logical switches. A VLAN is identified by a VLAN ID. The VLAN ID is usually a value from 1 to 4094, with 0 and 4095 reserved. On Cisco switches, VLAN 1 is commonly the default VLAN; this is not a universal default for every vendor or network. Every port on a switch can be assigned a different VLAN or a group of ports can be assigned a particular VLAN ID. VLANs allow network administrators to logically split a switch, allowing multiple broadcast domains to coexist on the same hardware while maintaining the isolation, security, and performance benefits of completely separate switches.

There are several advantages to creating VLANS:

- **Network segmentation:** Creating VLANs within a switch separates Layer 2 broadcast domains, but it does not automatically create a complete security boundary. Security between VLANs requires controls such as ACLs, firewall policy, and secure switch configuration. This kind of logical separation is very useful when creating networks for various departments in an enterprise is needed. Figure 17-10 shows three VLANs—VLAN 10, VLAN 20, and VLAN 30—each assigned to a different department.

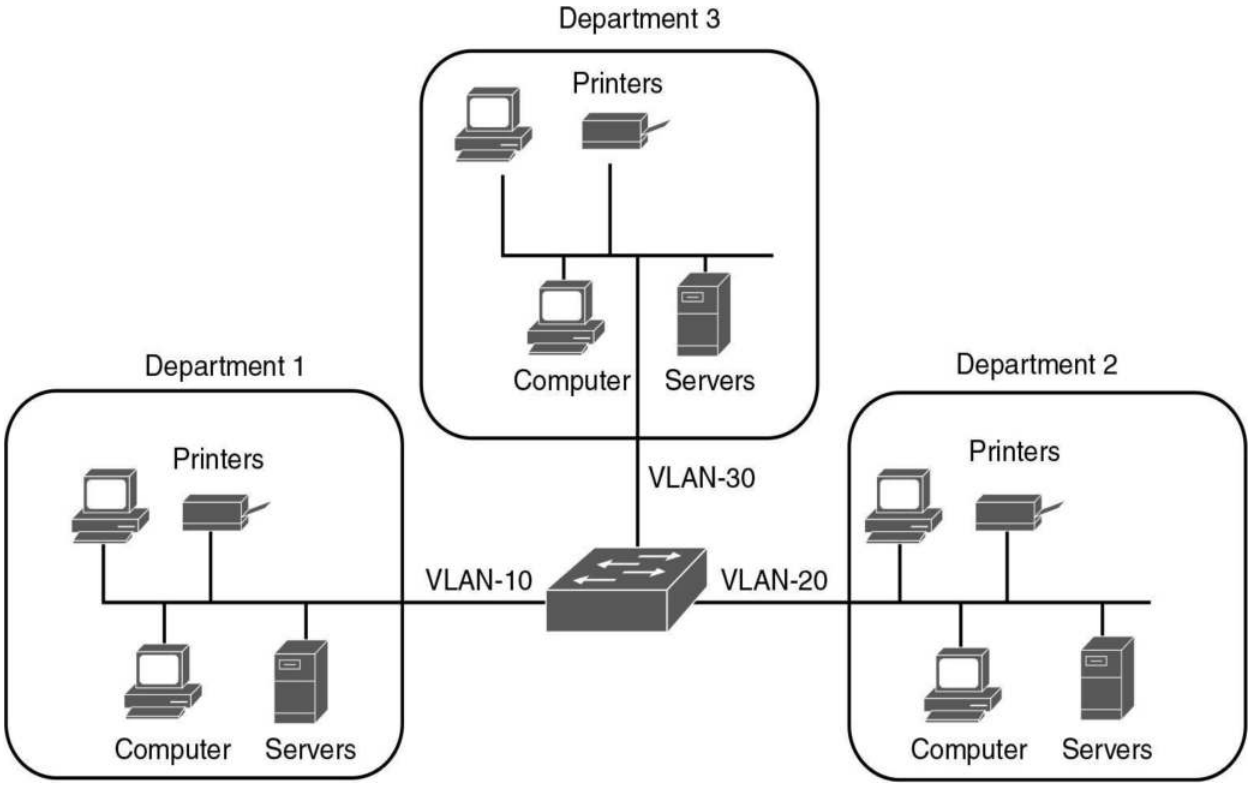


Figure 17-10 *Using a VLAN for Network Security*

- **Broadcast traffic distribution:** Segmenting a large LAN into smaller VLANs can reduce broadcast traffic because each broadcast packet is sent only to the relevant VLAN. For example, Figure 17-10 shows three VLANs—one for each domain—broadcast traffic will go to only the devices in the appropriate VLAN.
- **Performance increase:** Creating multiple broadcast domains reduces the broadcast traffic on the entire network tremendously, boosting the network's overall performance.

Say that you have a single switch. By default, all ports on this switch are in one VLAN, such as VLAN 1. Any port can be configured to be an access port or a trunk port:

- **Access port:** An access port is essentially a port that can be assigned to only one VLAN. You can change the port membership by specifying the new VLAN ID.
- **Trunk port:** A trunk port can have two or more VLANs configured. It can transport traffic for several VLANs simultaneously.

Figure 17-11 shows how trunk ports connect various VLANs across switches.

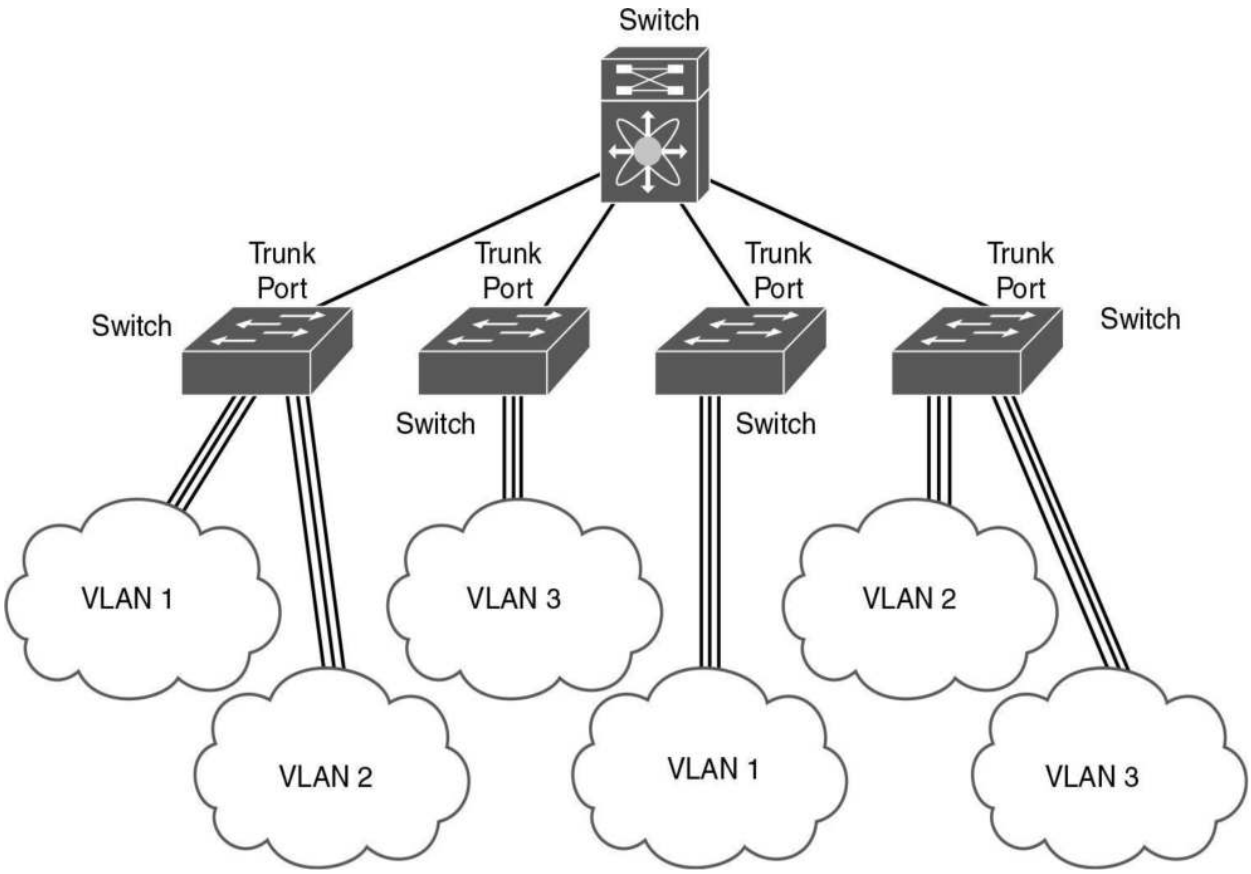


Figure 17-11 *VLAN Trunk Ports*

Routers

A router is a device that forwards packets between networks via the network layer of the OSI model (Layer 3). It forwards or routes packets based on the IP address of the destination device. A router also has the intelligence to determine the best path to reach a particular network or device. It can determine the next hop, or routing destination, by using routing protocols such as Routing Information Protocol (RIP), Open Shortest Path First (OSPF), and Border Gateway Protocol (BGP).

What is the difference between a Layer 3 switch and a router? Both forward packets based on IP addresses. Both can route between IP networks; the practical differences are usually platform role, interface types, feature set, and scale. Routers commonly provide a broader range of WAN, security, and service-provider features. Modern Layer 3 switches and routers can both use ASICs or other forwarding hardware while software builds and manages forwarding state.

Figure 17-12 shows a router connecting different types of networks.

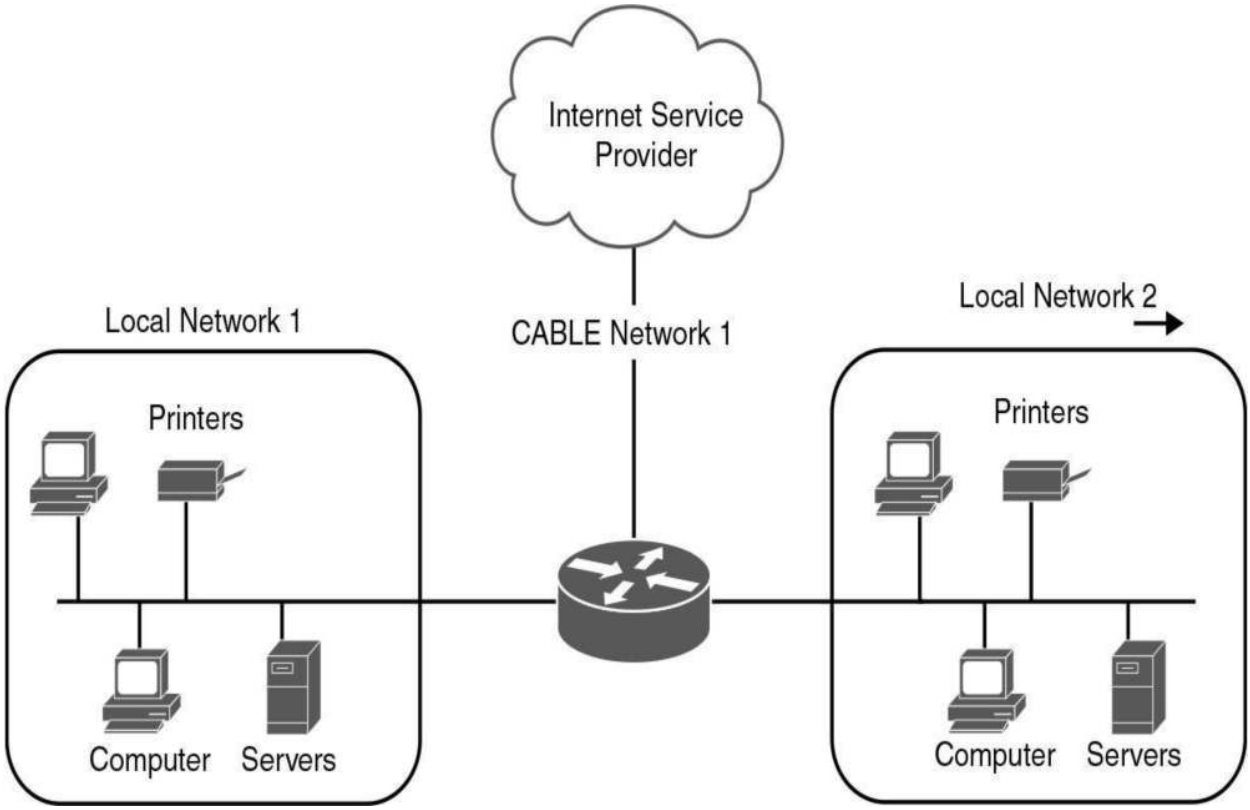


Figure 17-12 *Using a Router to Connect Different Networks*

Routing in Software

As discussed earlier in this chapter, routers participate in route discovery, path determination, and packet forwarding. Route discovery and path determination are part of the routing protocol, and as the router becomes part of this discovery process, it discovers and updates the routing tables accordingly. There are three types of forwarding:

- **Process switching:** The CPU is involved for every routed packet and requires a complete routing table lookup. Process switching, shown in Figure 17-13, is the slowest type of forwarding as each of the packets that the interface driver receives (step 1 in the figure) is punted and put in the input queue for the processor for further action (step 2). In this case, the processor receives the packets (step 3), determines the following interface it needs to send them to, rewrites the headers as needed, and puts the new packets in the output queue (step 4). Finally, the kernel driver for the new network interface picks up the packets and transmits them on the interface (step 5).

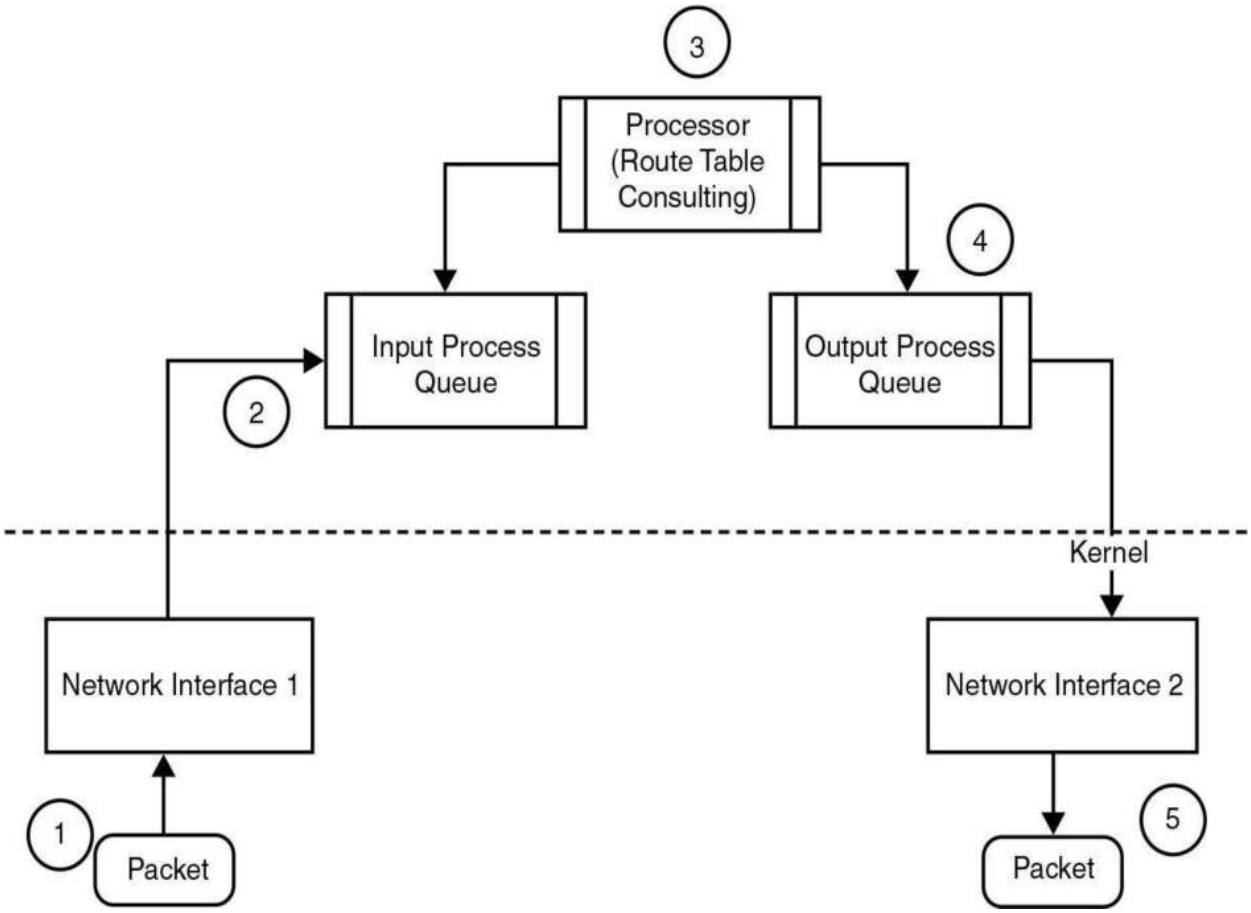


Figure 17-13 *Process Switching with the CPU Involved in Packet Routing Decisions*

- **Fast switching:** Fast switching is the next-level evolution of process switching. The CPU is involved for only the first packet that is routed. It determines the outgoing interface required for the packet and updates the route cache, which typically resides in the kernel. Once the route cache is populated, the input interface driver does a route-cache lookup and fast switches the packet to the outgoing interface. Figure 17-14 shows fast switching, in which the processor updates the route cache after learning the next hop. All subsequent packets are fast-switched.

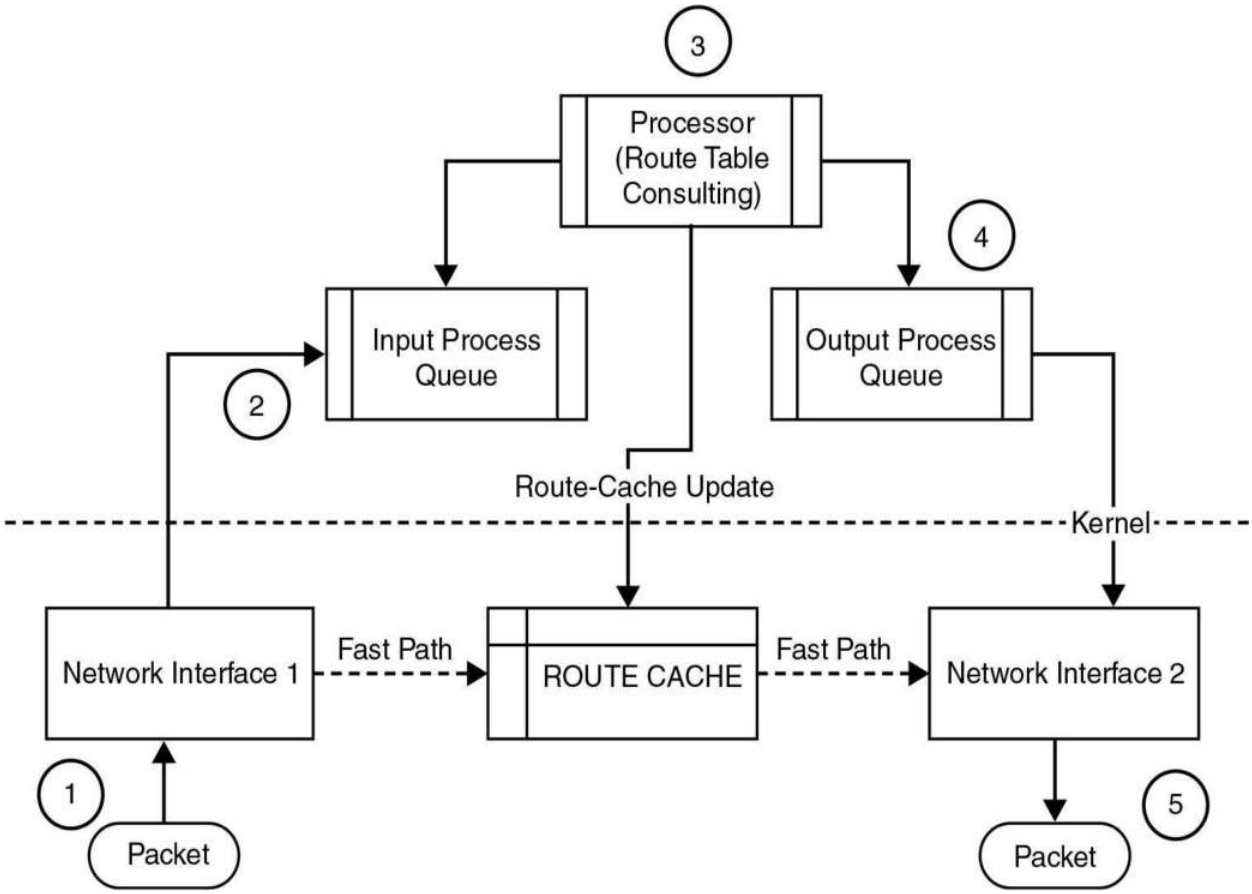


Figure 17-14 *Fast Switching and the Route Cache*

- **Cisco Express Forwarding (CEF) switching:** CEF switching is an advanced Cisco-proprietary form of fast switching. CEF switching sometimes involves specialized hardware and distributed routing, and the CPU may not need to get involved in the data path. It does this by building a Forwarding Information Base (FIB). The FIB contains destination prefixes and next-hop pointers for longest-prefix forwarding; a separate adjacency table contains outbound-interface and Layer 2 rewrite information. When a network topology or routing table change occurs, the change is also reflected in the FIB table. Figure 17-15 shows CEF switching, where Layer 2 and 3 information is used.

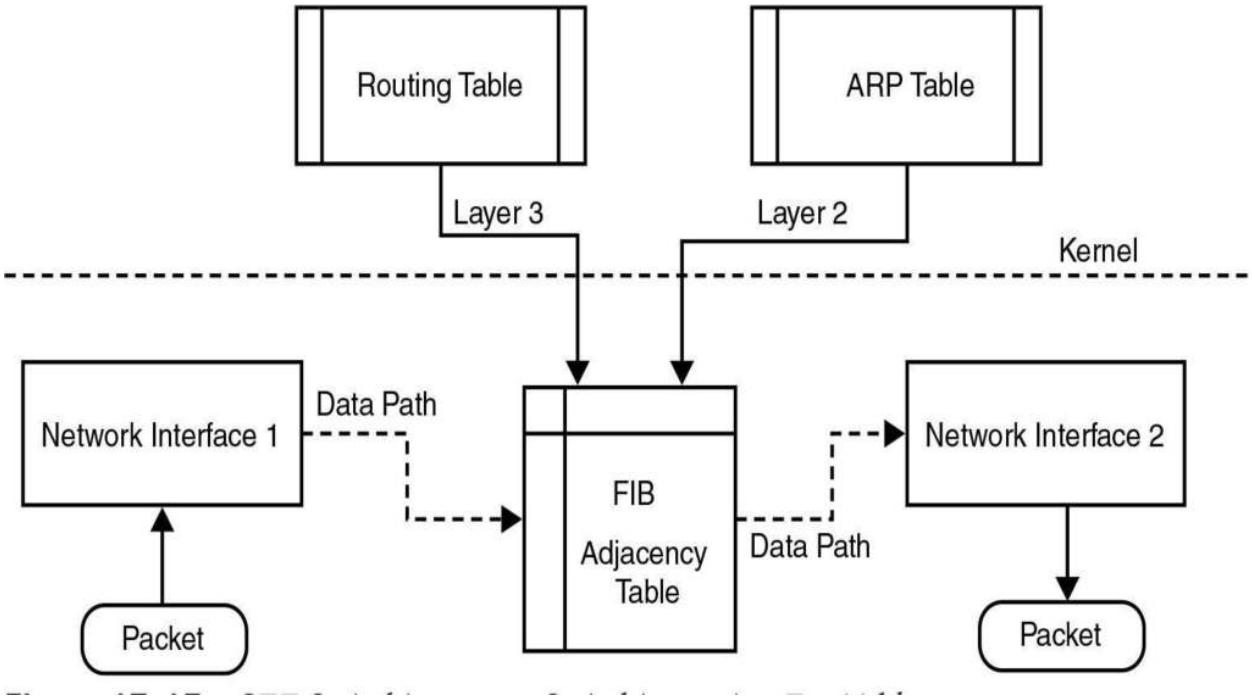


Figure 17-15 *CEF Switching: Data Switching Using FIB Tables*

This analogy is sometimes used to illustrate the three types of switching:

- Process switching is like doing math on paper: Write down each step and solve the problem.
- Fast switching, using the route cache, is like solving a problem by hand once and then simply recalling the answer from memory if the same problem is given again.
- CEF switching is like using formulas in an Excel spreadsheet, and when the numbers hit the cells, the answer is automatically calculated.

Functions of a Router

As discussed earlier in this chapter, a router is a device that appropriately and efficiently directs traffic from an incoming interface to the correct outgoing interface. Apart from this primary function, a router has several other functions:

- **Address translation:** A router uses Network Address Translation (NAT) to map private IP addresses to public IP addresses. Using NAT at the router conserves public IPv4 addresses, but NAT is not a substitute for a firewall or explicit security policy. Figure 17-16 shows address translation between private and public IPv4 address spaces.

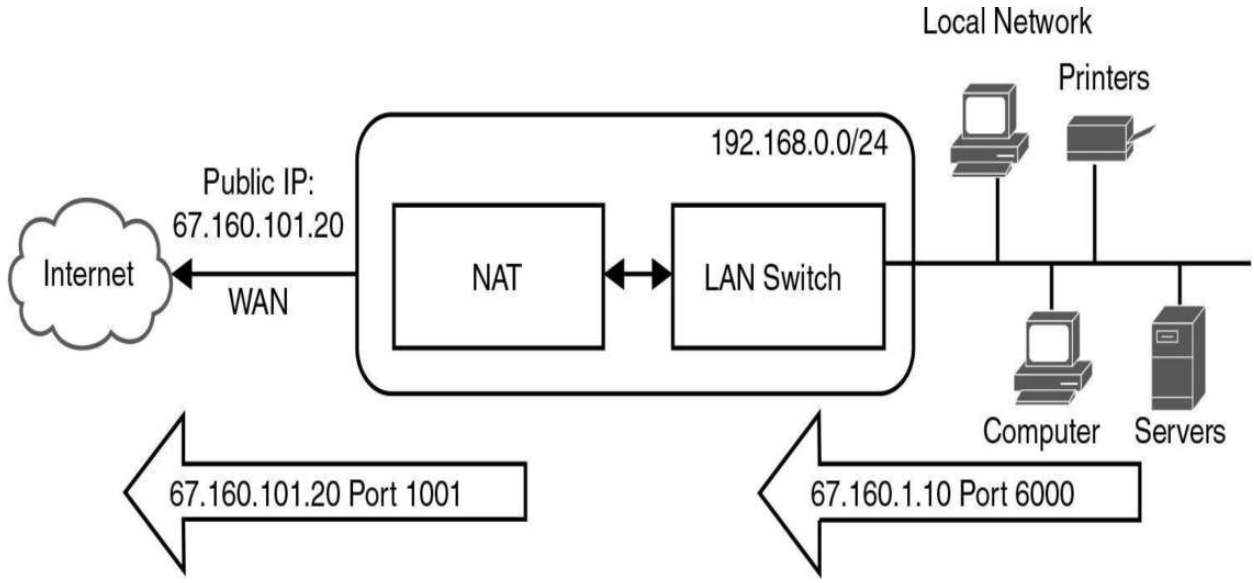


Figure 17-16 *Router NAT: Translating Internal Addresses to External Addresses*

- **IP address management:** A router uses Dynamic Host Configuration Protocol (DHCP) to assign IP addresses to devices that connect to the network. Clients send DHCPDISCOVER broadcast messages to determine if a local DHCP server exists in the network. If a DHCP server exists, it offers an IP address lease and configuration options such as the subnet mask, default gateway, DNS servers, and domain name in a DHCPOFFER message. The client can then send a DHCPREQUEST to the server for allocating an IP address. The server responds with a DHCPACK message, which may be broadcast, relayed, or unicast depending on the client's state and message fields, indicating that the IP address has been allocated and confirmed. Figure 17-17 shows a flow of clients requesting IP addresses via DHCP.

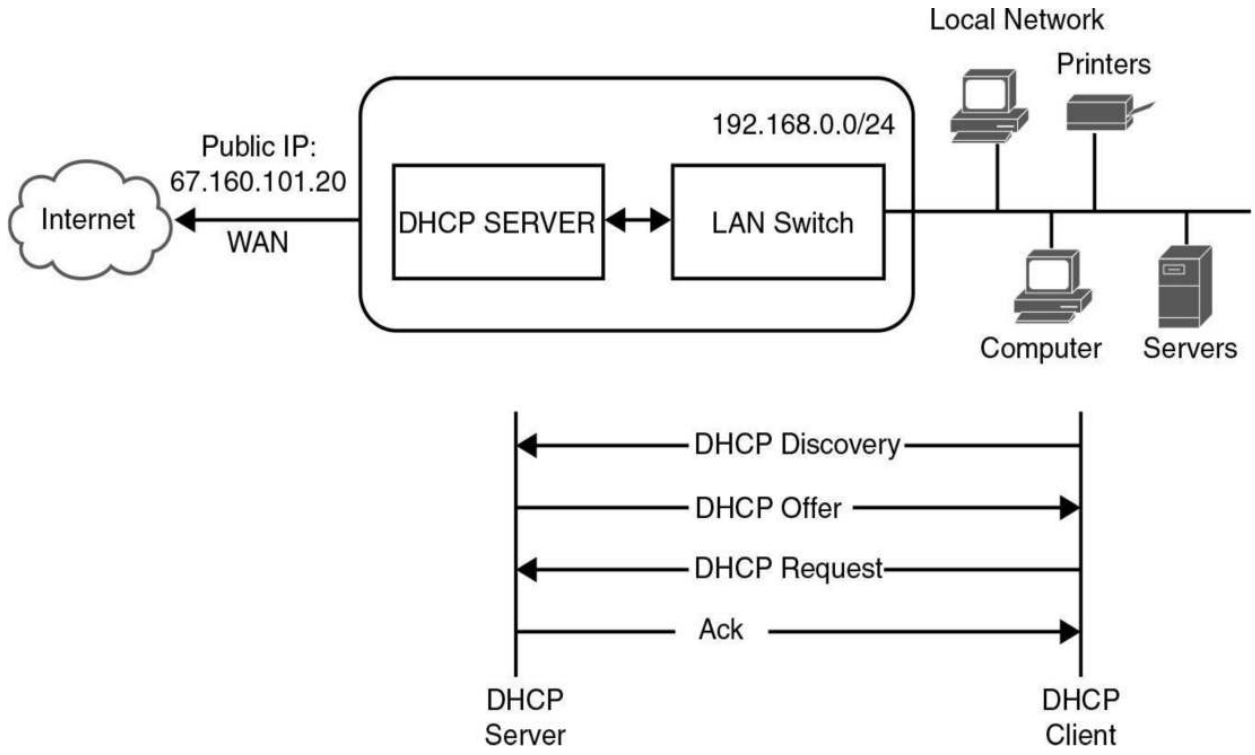


Figure 17-17 *Using DHCP to Assign IP Addresses*

- **Firewalls:** A firewall safeguards a network from intentional or unintentional intrusion. A firewall sits at the junction point or gateway between two networks—usually private and public networks such as the Internet. Users on the router's local network need to be protected from hackers and other malicious users. A firewall blocks traffic from unauthorized users. It also helps allow or block certain types of traffic on specific ports. For example, a firewall may only allow access to specific applications, such as HTTP or SSH, and block all UDP traffic. Figure 17-18 shows a firewall that will enable only HTTP access to the outside world and block all incoming UDP traffic.

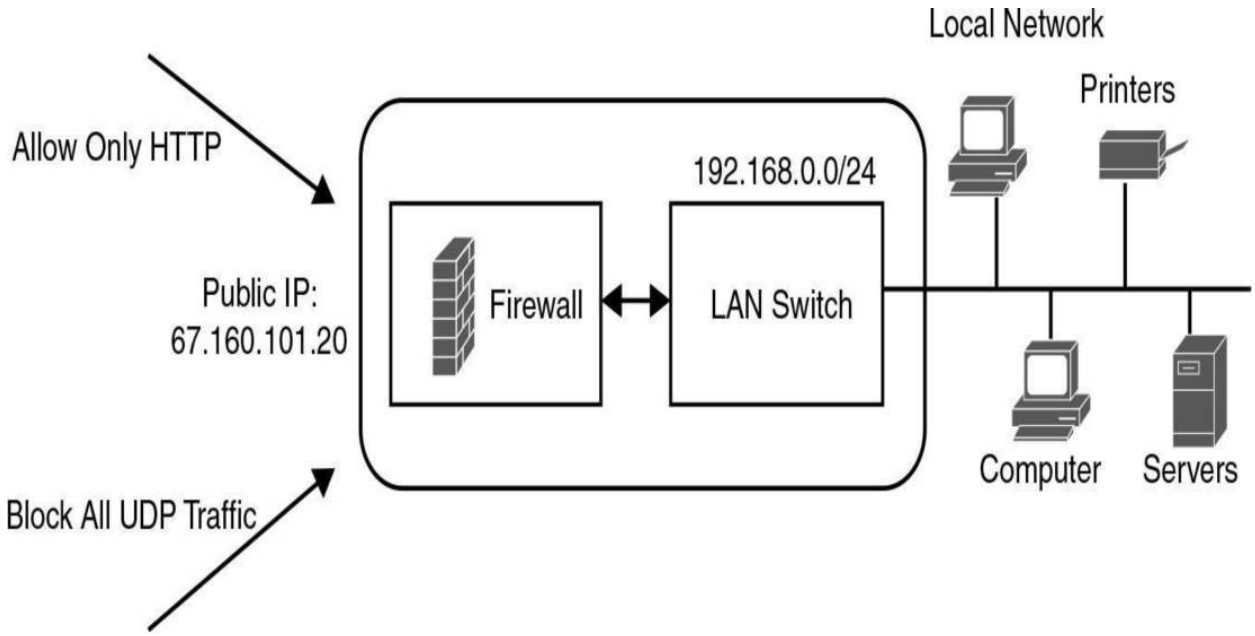


Figure 17-18 *Router Firewall: Allowing/Blocking Certain Apps or Traffic*

- **Domain name proxy:** Modern firewalls interact with cloud DNS servers such as Cisco Umbrella or Open DNS to resolve DNS queries. The queries and, in turn, the hosts can be stopped at the source if they are deemed harmful or malicious.

Network Diagrams: Bringing It All Together

So far in this chapter, you have learned about all the elements that constitute a network. The best way to visually represent a network topology is by using a network diagram. A network diagram maps out the structure of a network with different network element icons and connections between them. This kind of visual presentation makes it simple and easy for anyone to understand how a network is built and can help during initial network deployment and debugging.

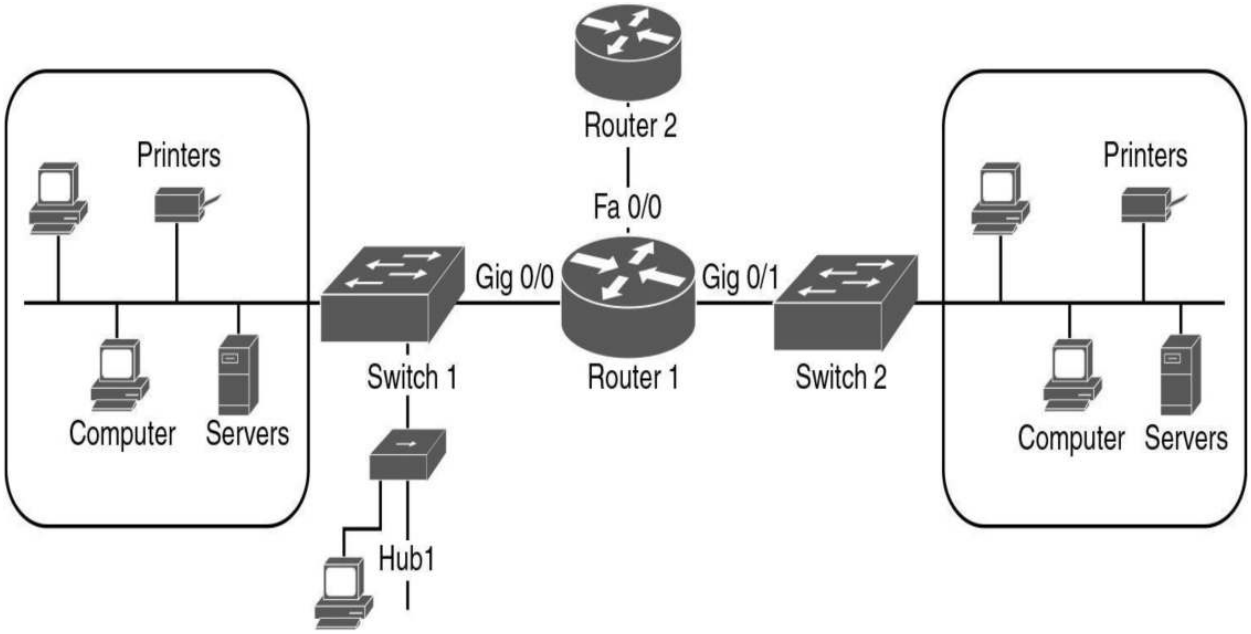


Figure 17-19 *Network Diagram*

Figure 17-19 shows a typical network topology. At the network's core is a router (Router 1) connecting three different networks. One router interface is connected to a switch (Switch 1), a second router interface is connected to a switch (Switch 2), and a third router interface is connected to a router (Router 2):

- Switch 1 connects to Gigabit-ether 0/0 on the router.
- Switch 2 connects to Gigabit-ether 0/1 on the router.
- Router 2 connects to Fast-ethernet 0/0 on the router.

Switch 1 and Switch 2 have networks consisting of various devices such as computers, printers, and so on. Router 2 is the Internet router with firewall functionality and a connection to the service provider.

Software-Defined Networking

The term software-defined networking (SDN) applies to a type of network architecture design that enables IT managers and network engineers to manage, control, and optimize network resources programmatically.

SDN decouples network configuration and data flow engineering, regardless of the underlying hardware infrastructure. It allows you to consistently control the network by using standard open APIs.

Networks are often shown using a three-level architecture, as shown in Figure 17-20, which consists of hardware and two planes:

- **Data plane:** As described earlier in this chapter, a router can route packets faster by using techniques such as fast switching or CEF switching. These techniques for punting packets from the incoming interface to the outgoing interface operate on what is traditionally known as the data plane. The main objective of the data plane is to determine how the incoming packet on a port must be forwarded to an outgoing port based on specific values in the packet headers.
- **Control plane:** Routing protocols and other protocols make up the control plane. The control plane determines how a packet is routed among routers or other network elements as the packet traverses end-to-end from the source host to the destination host. The control plane also deals with packets destined for the router. The separate management plane supports device administration, configuration, monitoring, and access. Management functions include initializing interfaces with default configurations, IP addresses, policies, user accounts, etc.

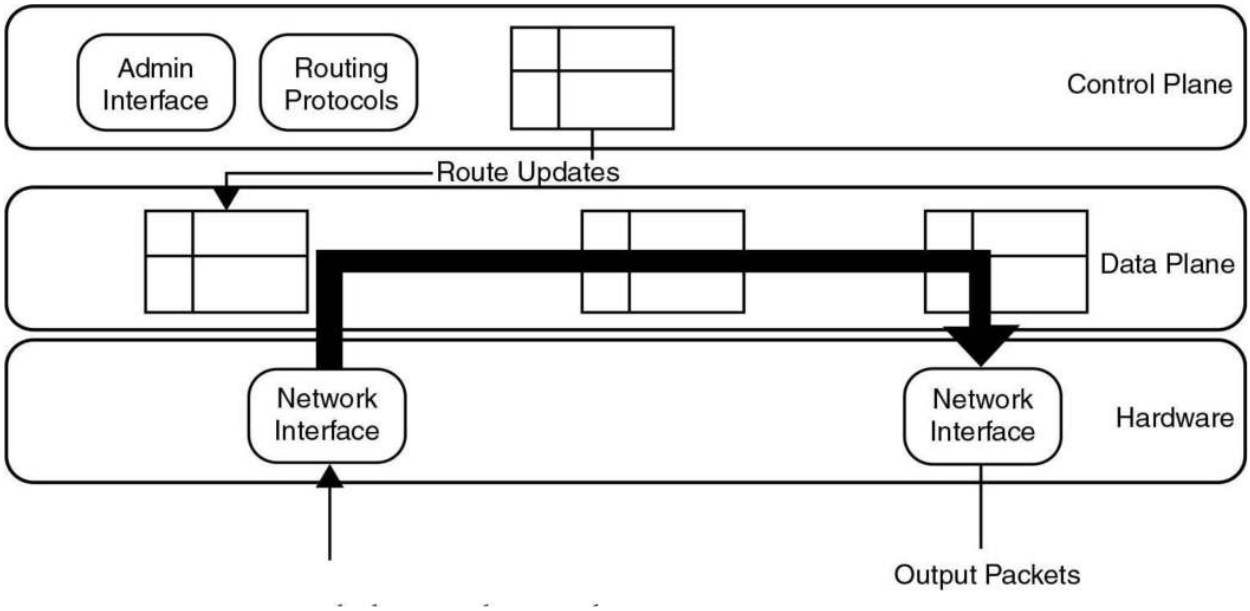


Figure 17-20 *Control Plane and Data Plane*

Figure 17-21 shows three network elements: a router, a switch, and a wireless access point. For each of them, the figure shows the control and data planes. Managing these devices includes the functions you use to control and monitor these devices. As the controlling and monitoring function has to be done on each device, we call this architecture a "distributed control plane architecture." The drawback is that network engineers or administrators must manage each device independently, logging in to each device and setting it up. Imagine that you have to make a simple configuration change, such as enabling a syslog server; you would have to log in to each device independently and make the changes.

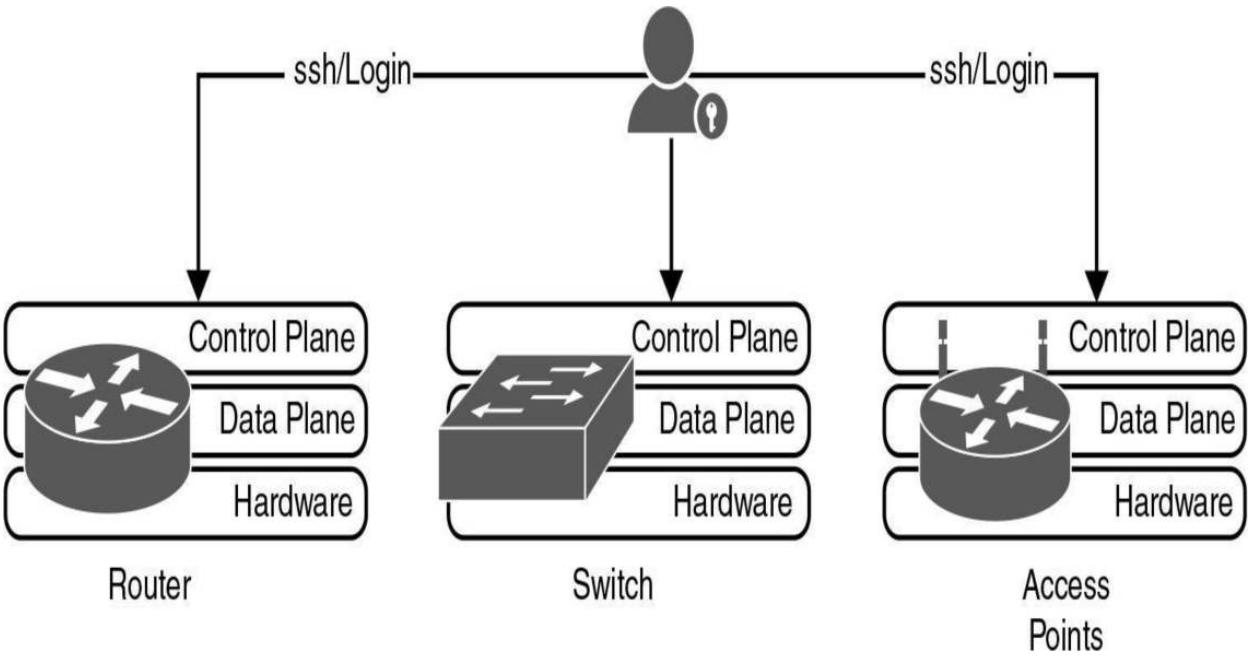


Figure 17-21 *Distributed Control Plane Architecture*

A network controller can be used to prevent the inefficiency of manually managing network devices. A network controller provides logically centralized policy and programmability; depending on the architecture, devices may retain local control-plane protocols and agents. The network controller can then communicate with the network elements, and the network elements can send management data to the controller. This is commonly described as a logically centralized control plane even when the controller service is distributed for scale and resiliency. Figure 17-22 shows how a network controller can programmatically manage and control various networking elements using the northbound and southbound APIs.

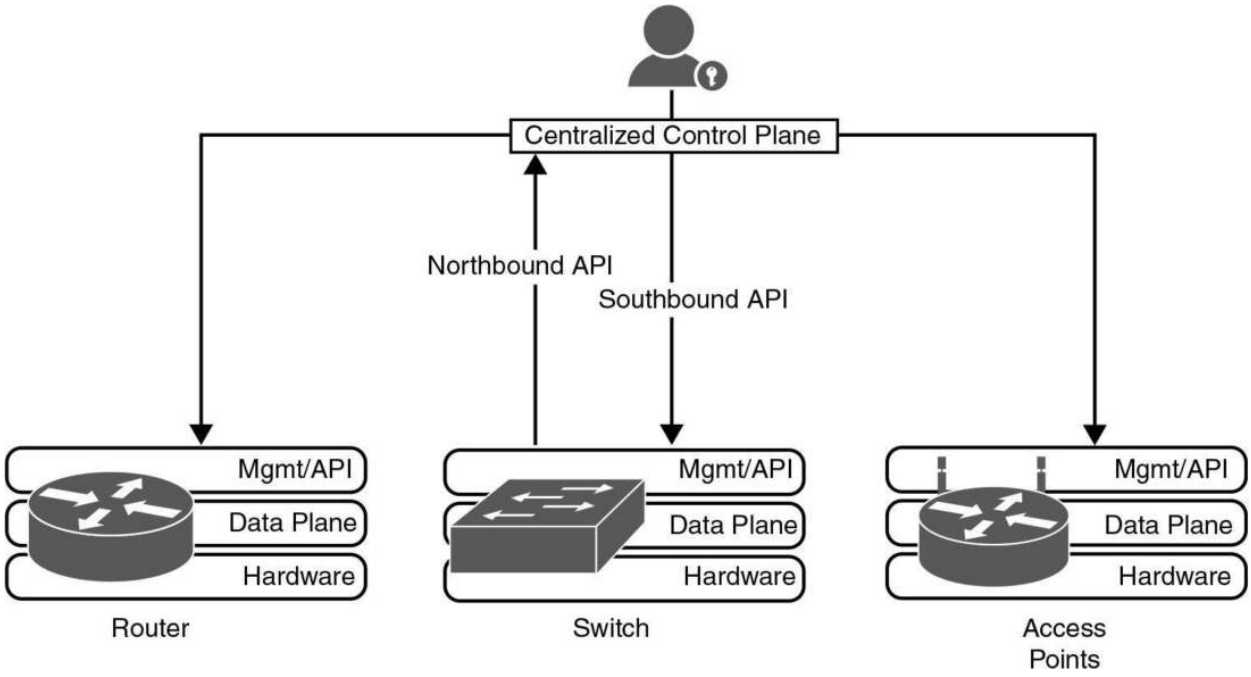


Figure 17-22 *Centralized Control Plane with a Network Controller*

Communication between the network controller and the network elements is bidirectional. Both sides use application programming interfaces (APIs), which allow the network controller and the network devices to communicate programmatically. The network controller interacts with the network devices via southbound APIs, and applications or orchestration systems interact with the controller via northbound APIs.

SDN Controllers

An SDN controller is often considered the "brain" in a modern network. The Open Networking Foundation (ONF) defines SDN as follows:

> an emerging architecture that is dynamic, manageable, cost-effective, and adaptable, making it ideal for the high-bandwidth, dynamic nature of applications. This architecture decouples the network control and forwarding functions, enabling the network control to become directly programmable and the underlying infrastructure to be abstracted for applications and network services.

An SDN controller possesses a "global" view of the entire network. It knows about all the network elements that constitute the network, the best paths between them, and other potential routes. As indicated in the ONF SDN definition, SDN is a network architecture that separates the control and data planes for network devices and provides centralized management. Two commonly discussed SDN technologies are:

- **OpenFlow:** An ONF-managed protocol for communication between an SDN controller and the forwarding plane of managed network devices.
- **OpenDaylight:** A Linux Foundation open-source SDN controller platform that supports multiple southbound protocols, including OpenFlow, NETCONF, BGP-LS, PCEP, and OVSDB.

As shown in Figure 17-23, SDN involves three layers: application layer, control layer, and infrastructure layer.

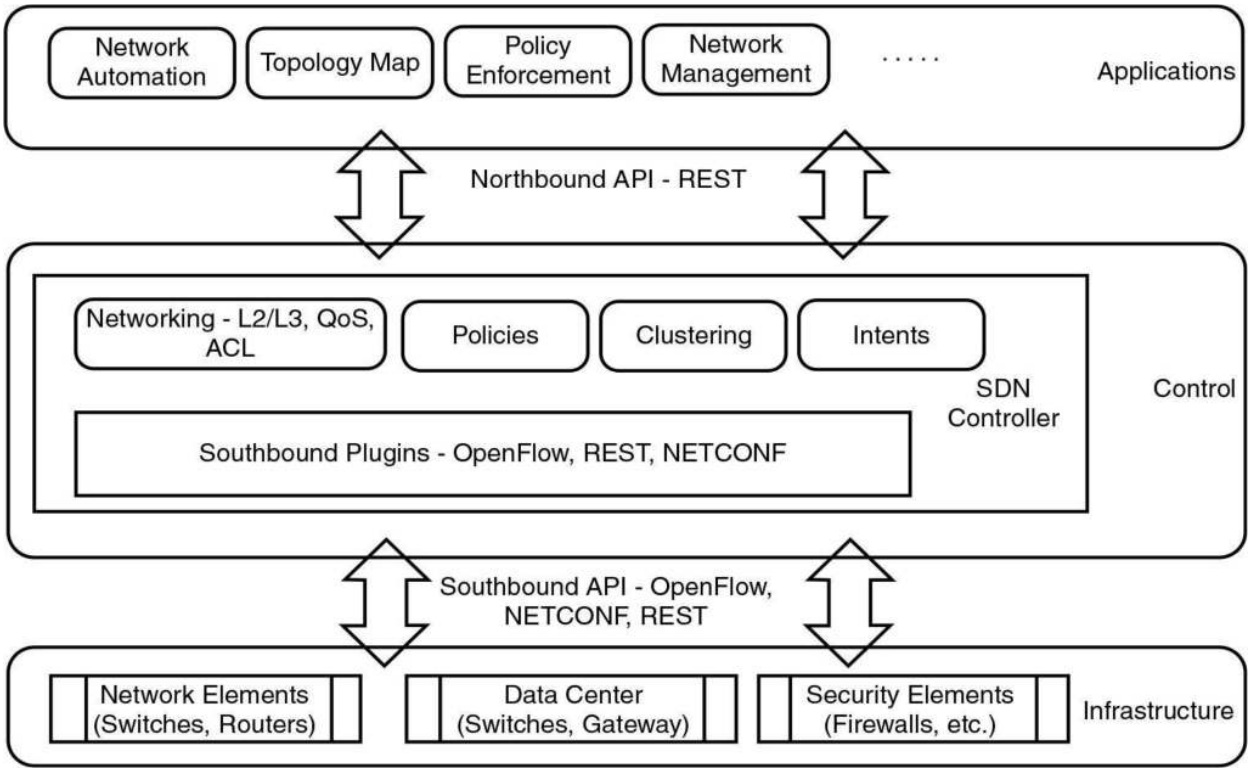


Figure 17-23 *Software-Defined Networking: Layered Architecture*

The infrastructure layer is composed of networking equipment and elements that form the actual network and elements that help to forward network traffic. It could be a set of network switches and routers in the network or data centers. The infrastructure layer is the physical layer over which network virtualization functions lay via the control layer.

The control layer is where the SDN controllers reside to control network infrastructure. The control layer has the business logic to fetch and maintain different types of network information, state details, topology details, and statistics details. The SDN controller is all about managing networks, and it has all the control logic for network use cases such as switching, routing, Layer 2 VPNs, Layer 3 VPNs, firewall security rules, DNS, DHCP, and clustering. Cisco implements various services in its SDN controllers. These services expose APIs (typically REST-based) to the upper layer (application layer), which makes life easy for network administrators who use apps on top of SDN controllers to configure, manage, and monitor the underlying network. The control layer exposes two types of interfaces:

- **Northbound interface:** This interface is used for communication with the upper layer (the application layer) and is in general realized through REST APIs of SDN controllers.
- **Southbound interface:** This interface is meant for communication with the lower layer (the infrastructure layer) of network elements and is in general, realized through southbound protocols, such as OpenFlow and NETCONF.

The application layer is a developing area where many innovations are happening. The applications in this layer leverage information about the network topology, network state, network statistics, and so on. Several types of applications can be developed, such as those related to network automation, network configuration and management, network monitoring, network troubleshooting, network policies, and security. Such SDN applications can provide end-to-end solutions for real-world enterprise and data center networks.

Cisco Software-Defined Networking (SDN)

Cisco has introduced many SDN controllers in various domains, including the data center, campus, and service provider domains. The following is the list of Cisco's SDN products:

- **Cisco Application Centric Infrastructure (ACI):** ACI was the first Cisco SDN solution, and it has three components:
- **Application Network Profile:** This is a collection of endpoint groups (EPGs), their connections, and the policies that define the connections.
- **Application Policy Infrastructure Controller (APIC):** This centralized software controller manages downstream switches and acts as a management plane.
- **ACI fabric:** This is the connection between spine and leaf switches. In the ACI world, the spine and leaf are the Cisco Nexus 9000 Series switches, which act as the control plane and the data plane of the ACI.
- **Cisco Digital Network Architecture (DNA):** Cisco DNA introduces the concept of intent-based networking, which interprets business needs into technical solutions.

- **Cisco Network Services Orchestrator (NSO):** This solution mostly automates tasks in service provider environments. NSO is a multivendor controller installed in Linux that supports NETCONF, OpenFlow, SNMP, and APIs.
- **Cisco Software-Defined WAN (SD-WAN):** SD-WAN deals with creating and managing WAN connections in a cloud-based environment. SD-WAN solutions have several key features—including segmentation, centralized policies, zero-touch provisioning, and configuration templates—that can be very helpful to customers. Cisco SD-WAN has four main components, each with a very specific role:
- **Cisco SD-WAN Manager (formerly vManage) (for management):** Cisco SD-WAN Manager is a GUI-based network management system that handles the management plane. Cisco SD-WAN Manager is a single pane of glass that provides various key stats. An operations team can use Cisco SD-WAN Manager for day-to-day operational activities.
- **Cisco SD-WAN Controller (formerly vSmart):** The SD-WAN Controller manages the control plane. It performs path calculation and route advertisement, allowing the data plane to focus on packet forwarding.
- **WAN Edge router (including legacy vEdge platforms):** A WAN Edge router forwards packets based on policies distributed by the Cisco SD-WAN Controller. It maintains control connections to receive updates.
- **Cisco SD-WAN Validator (formerly vBond):** The SD-WAN Validator authenticates and authorizes WAN Edge routers joining the fabric and helps them discover Cisco SD-WAN Controllers and Managers.

Exam Preparation Tasks

As mentioned in the section "How to Use This Book" in the Introduction, you have a couple of choices for exam preparation: the exercises here, Chapter 19, "Final Preparation," and the exam simulation questions on the companion website.

Review All Key Topics

Review the most important topics in this chapter, noted with the Key Topic icon in the outer margin of the page. Table 17-3 lists these key topics and the page number on which each is found.

Table 17-3 *Key Topics*

Key Topic Element	Description	Page
Paragraph	Star, bus, and ring network topologies	
Paragraph	Hubs	
Paragraph	Layer 2 and Layer 3 switches	
Paragraph	VLANs	
List	Methods of routing packets in a router	
Paragraph	SDN controllers	

Define Key Terms

Define the following key terms from this chapter and check your answers in the glossary:

local-area network (LAN)
campus-area network (CAN)
metropolitan-area network (MAN)
wide-area network (WAN)
virtual LAN (VLAN)
Cisco Express Forwarding (CEF)
Forwarding Information Base (FIB) table
Network Address Translation (NAT)
Dynamic Host Configuration Protocol (DHCP)
software-defined networking (SDN)

Chapter 18. IP Services

This chapter covers the following topics:

- **Common Networking Protocols:** This section introduces protocols that are commonly used in networks and that you should be familiar with.
- **Layer 2 Versus Layer 3 Network Diagrams:** This section covers different types of network diagrams.
- **Troubleshooting Application Connectivity Issues:** This section describes how to troubleshoot application connectivity issues.

This chapter covers IP services concepts. It starts with an introduction to common networking protocols that you are bound to use on a daily basis and then discusses them in detail:

- Dynamic Host Configuration Protocol (DHCP) is used to dynamically allocate IP configuration data to network clients so that the clients can get access to the network.
- Domain Name System (DNS) is a hierarchical naming system used mostly to resolve domain names to IP addresses.
- Network Address Translation (NAT), while not a protocol per se, deals with translations between private IP networks and public, globally routable IP networks.
- Simple Network Management Protocol (SNMP) has been developed for remote network monitoring and configuration.
- Network Time Protocol (NTP) is used to synchronize date and time between devices on a network.

The chapter also provides a comparison between Layer 2 and Layer 3 network diagrams, as well as a troubleshooting scenario for application connectivity issues.

"Do I Know This Already?" Quiz

The "Do I Know This Already?" quiz allows you to assess whether you should read this entire chapter thoroughly or jump to the "Exam Preparation Tasks" section. If you are in doubt about your answers to these questions or your own assessment of your knowledge of the topics, read the entire chapter. Table 18-1 lists the major headings in this chapter and their corresponding "Do I Know This Already?" quiz questions. You can find the answers in Appendix A, "Answers to the 'Do I Know This Already?' Quiz Questions."

Table 18-1 *"Do I Know This Already?" Section-to-Question Mapping*

Foundation Topics Section	Questions
Common Networking Protocols	1–5
Layer 2 Versus Layer 3 Network Diagrams	6
Troubleshooting Application Connectivity Issues	7

Caution

The goal of self-assessment is to gauge your mastery of the topics in this chapter. If you do not know the answer to a question or are only partially sure of the answer, you should mark that question as wrong for purposes of self-assessment. Giving yourself credit for an answer that you correctly guess skews your self-assessment results and might provide you with a false sense of security.

1. What are some of the benefits that DHCP offers? (Choose two.)
   - **a.** Reduced network endpoint configuration tasks and costs
   - **b.** Elimination of single points of failure for network parameter configuration
   - **c.** Decentralized management of network configuration

d. Centralized management of network parameter configuration

2. What DNS component is configured on a network client?

   a. Root server

   b. TLD server

   c. DNS resolver

   d. TLC server

3. What are some of the benefits of Network Address Translation? (Choose three.)

   a. Conserving public IP address space

   b. Increased security due to hiding of internal addressing

   c. Loss of end-to-end traceability and visibility

   d. Loss of end-to-end functionality

   e. Flexibility when changing ISPs

4. What version of SNMP was developed with a security focus in mind?

   a. SNMPv3

   b. SNMPv1

   c. SNMPv2c

   d. SNMPv4

5. What transport layer protocol and port are used by NTP?

   a. TCP 123

   b. TCP 120

   c. UDP 120

   d. UDP 123

6. What network devices should be included in a Layer 3 diagram?

   a. Switches, routers, and firewalls

   b. Routers, firewalls, and load balancers

   c. Switches, firewalls, and load balancers

   d. Bridges, switches, and hubs

7. What popular network utility is used to troubleshoot DNS issues?

   a. traceroute

   b. dnslookup

   c. nslookup

   d. ping

Foundation Topics

Common Networking Protocols

The following sections cover common networking protocols that network engineers and software developers alike should be familiar with. Knowledge of these protocols and technologies will give you better insight into how networks interact with applications to be able to offer network clients an optimized and seamless experience.

Dynamic Host Configuration Protocol (DHCP)

Dynamic Host Configuration Protocol (DHCP), as the name suggests, is a protocol used for dynamically configuring hosts with network connectivity information. For any host device connected to a network to be able to send and transmit data, it needs to have network parameters such as IP address, subnet, default gateway, and DNS servers configured. This configuration can be done either manually or automatically, using protocols such as DHCP. Manual configuration of network parameters for hosts on a network is time-consuming and prone to errors; therefore, it is not very often implemented in real-world networks anymore. DHCP is extensively used for automatically distributing network configuration parameters to all network endpoints, including end-user devices and network devices. For networks that have already migrated to IP version 6 (IPv6) or that are running dual-

stack IP version 4 (IPv4) and IPv6 addressing, DHCPv6 and the IPv6 autoconfiguration option can be used for dynamic and automatic network parameter assignment. IPv6 autoconfiguration can be used to quickly and dynamically assign IPv6 addresses to network clients, and DHCPv6 is used to assign not just IPv6 addresses but also DNS servers and domain names.

DHCP has two components: a protocol for delivering network device configuration information from a DHCP server to a network host and a mechanism for allocating that configuration information to hosts. DHCP works using a client/server architecture, with a designated DHCP server that allocates IP addresses and network information and that delivers that information to DHCP clients that are the dynamically configured network endpoints.

Besides basic network connectivity parameters such as IP addresses, subnet masks, default gateways, IP addresses of DNS servers, and local domain names, DHCP also supports the concept of options. With DHCP options, a DHCP server can send additional configuration information to its clients. For example, Cisco IP Phones use option 150 provided by the DHCP server to obtain IP addresses of the TFTP servers that hold configuration files for the IP Phones; Cisco wireless access points use DHCP option 43 to obtain the IP address of the Cisco wireless LAN controller that they need to connect to for management purposes.

DHCP for IPv4 is defined and described in RFC 2131: *Dynamic Host Configuration Protocol* and RFC 2132: *DHCP Options and BOOTP Vendor Extensions*. For IPv6, DHCPv6 is currently specified by RFC 9915: *Dynamic Host Configuration Protocol for IPv6 (DHCPv6)*. RFC 9915 consolidates stateful and stateless configuration and prefix delegation and obsoletes RFC 8415, which had obsoleted RFCs 3315, 3633, and 3736.

Some of the benefits of using DHCP instead of manual configurations are

- **Centralized management of network parameters configuration:** DHCP servers usually manage the network configuration settings for several subnets and represent the central source of truth and the configuration point for all dynamic network parameters needed for network endpoints to be able to connect to the network. This makes it much easier to manage network address assignment compared to using several disparate systems or Excel files.
- **Reduced network endpoint configuration tasks and costs:** Dynamically allocating network connectivity information brings huge cost and time savings compared to manually performing the same tasks. This is especially true in medium to larger enterprise network environments and for Internet service providers (ISPs). Imagine ISPs needing to send out technicians to perform manual network changes to all of their customers when they start using their service every day. By using DHCP and dynamic network address configuration, client modems can be configured within seconds, without any manual intervention.

DHCP is built on top of a connectionless service model using User Datagram Protocol (UDP). DHCP servers listen on UDP port 67 for requests from the clients and communicate with the DHCP clients on UDP port 68. There are several ways network configuration information is allocated by the DHCP server:

- **Automatic allocation:** With automatic allocation, the DHCP server assigns a permanent IP address to the client.
- **Dynamic allocation:** With dynamic allocation, the DHCP server assigns an IP address to the client for a limited period of time called the *lease time*.
- **Manual allocation:** With manual allocation, an administrator preassigns an address to a client identifier or hardware address, and the DHCP server delivers that predetermined address and configuration options.

As mentioned previously, DHCP defines a protocol and a process for how to assign network configuration information to devices connecting to the network. The process defines the methodology used to configure the DHCP server. Usually, a DHCP server serves one or more client subnets. Once the client subnet is defined, a pool of addresses from that subnet is configured as available addresses for client allocation. Additional information such as subnet mask, the IP address of the default gateway, and DNS servers is the same for the whole subnet, so these configuration parameters apply to the whole subnet rather than to each end host device. In some cases, the DHCP client and the server are located in different subnets. In such a case, a DHCP relay agent can be used to relay the DHCP packets between clients and servers. Any host on the network can act as a relay agent, but in most cases, the default router for the client subnet acts as a DHCP relay agent.

Forwarding DHCP messages between clients and servers by the relay agents is different from regular routing and forwarding. While regular forwarding is transparent for the endpoints involved in the exchange of data, with DHCP forwarding, DHCP relay agents receive inbound DHCP messages on the client interface and generate new DHCP messages on the interface connecting to the server. The DHCP relay agent effectively becomes an on-path intermediary for the DHCP traffic between clients and servers. Figure 18-1 illustrates DHCP relay agent functionality.

Figure 18-1 *DHCP Relay Agent*

Key Topic

DHCP operations fall into the following four phases (see Figure 18-2):

- Server discovery
- Lease offer
- Lease request
- Lease acknowledgment

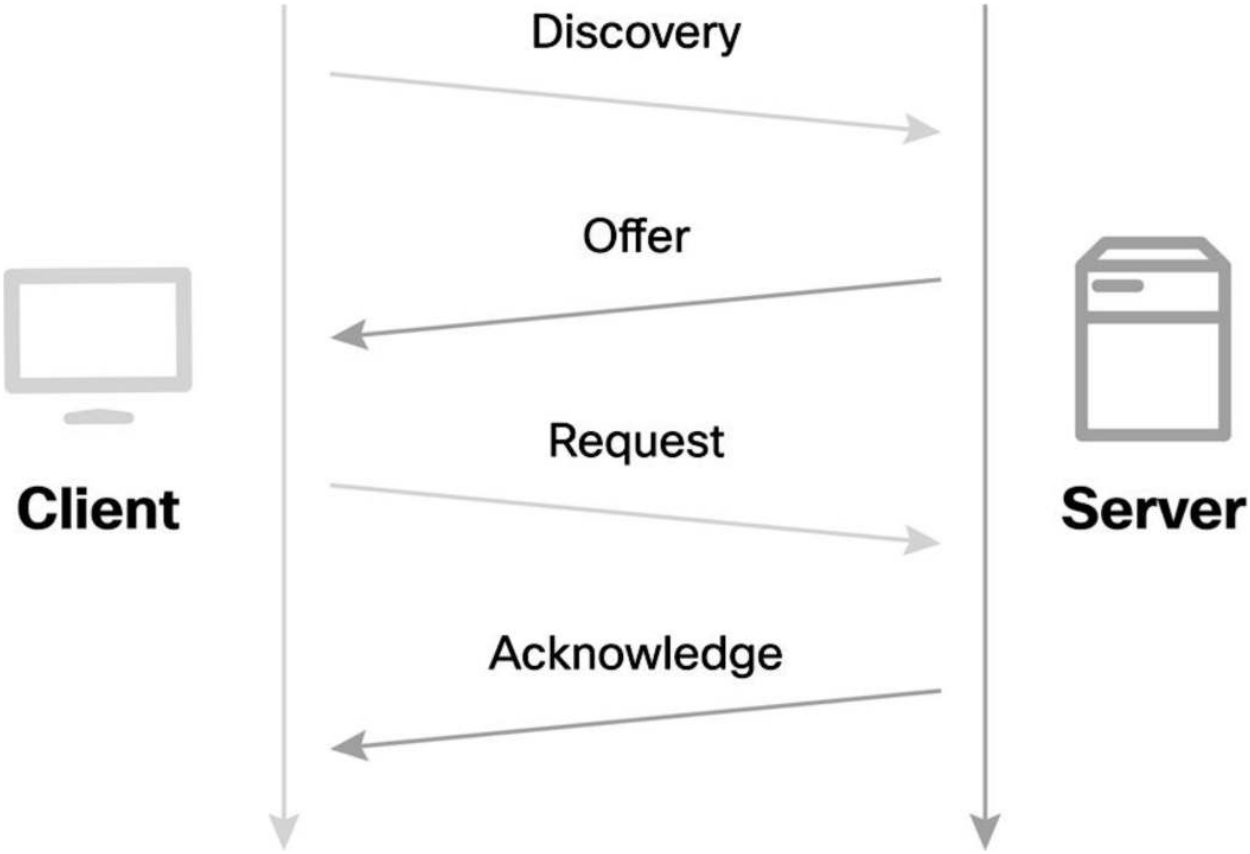


Figure 18-2 *DHCP State Machine*

Server Discovery

When a client first boots up and comes online on the network, it broadcasts a DHCPDISCOVER message with a destination address of the all-subnets broadcast address (255.255.255.255) with a source address of 0.0.0.0 because the client doesn't have any IP address at this stage. If a DHCP server is configured for this subnet, it receives the all-subnets broadcast message and responds with a DHCPOFFER message containing the network configuration parameters for the client. If a DHCP server isn't directly configured on the same subnet as the clients but there is a DHCP relay agent, the agent forwards the request to the DHCP server and will also forward the server offer to the requesting client.

Lease Offer

A DHCP server that receives a DHCPDISCOVER message from a client responds on UDP port 68 with a DHCPOFFER message addressed to that client. The DHCPOFFER message contains initial network configuration information for the client. Several fields in the DHCPOFFER message are of interest for the client:

- **chaddr:** This field contains the MAC address of the client to help the client know that the received DHCPOFFER message is indeed intended for it.
- **yiaddr:** This field contains the IP address assigned to the client by the server.
- **options:** This field contains the associated subnet mask and default gateway. Other options that are typically included in the DHCPOFFER message are the IP address of the DNS servers and the IP address lease and renewal time.

After the client receives a DHCPOFFER message, it starts a timer and waits for further offers from other DHCP servers that might serve the same client subnet.

Lease Request

After the client has received the DHCPOFFER from the server, it responds with a DHCPREQUEST message that indicates its intent to accept the network configuration information contained in the DHCPOFFER message. The client moves to the Request state in the DHCP state machine. Because multiple DHCP servers might be serving the same client subnet, the client might receive multiple DHCPOFFER messages, one from each DHCP server that received the DHCPDISCOVER message. The client chooses one of the DHCPOFFER messages; in most implementations of the DHCP client, this is the first DHCPOFFER received. The client replies to the server with a DHCPREQUEST message. The DHCP server chosen is specified in the Server Identifier option field of the DHCPREQUEST message. The DHCPREQUEST message has the all-subnets broadcast address as the destination address once more, so all DHCP servers receive this message and can determine whether the client accepted their offer. The source IP address of the DHCPREQUEST message is still 0.0.0.0 because the client has not yet received a confirmation from the DHCP server that it can use the offered IP address.

Lease Acknowledgment

The DHCP server receives the DHCPREQUEST message from the client and acknowledges it with a DHCPACK message that contains the IP address of the DHCP server and the IP address of the client. Depending on the client's state and message fields, the DHCPACK may be broadcast, relayed, or unicast. Once the client receives the DHCPACK message, it becomes bound to the IP address and can use it to communicate on the network. The DHCP server stores the IP address of the client and its lease time in the DHCP database.

Releasing

A DHCP client can relinquish its network configuration lease by sending a DHCPRELEASE message to the DHCP server. The lease is identified by using the client identifier, the chaddr field, and the network address in the DHCPRELEASE message.

Domain Name System (DNS)

Domain Name System (DNS) is a directory of networks that maps names of endpoints to IP addresses. DNS performs a critical role in all networks and especially on the Internet. It is much easier to remember the website www.cisco.com than it is to remember the IP address to which it resolves—173.37.145.84 for IPv4 or 2600:1408:2000:1b3:0:0:0:b33 for IPv6. DNS is responsible for the process of resolving a hostname to an IP address. Each host endpoint and any device connecting to the network need to have an IP address configured to be able to communicate on the network. The IP address is like a street address, because every device on the Internet can be located based on its IP address. For example, when a user loads a web page, a translation must happen between the website name (cisco.com) and the machine-friendly IP address needed to locate that web page. This process is abstracted from the user because the user doesn't need to know what is happening in the background with the resolution of names into IP addresses.

The DNS resolution process has several critical components:

- The DNS recursive resolver is the server that receives DNS queries from client machines and makes additional requests to resolve the client query.
- Root name servers at the top of the DNS hierarchy are the servers that have lists of the *top-level domain (TLD) name servers*. They are the first step in resolving hostnames to IP addresses.
- TLD name servers host the last portion of a hostname. For example, the TLD server in the cisco.com example has a list for all the .com entries. All the other domains have TLD servers as well (.net, .org, and so on).
- The authoritative name server is the final step in the resolution process. It is the authoritative server for that specific domain. In the case of cisco.com, there are three authoritative servers: ns1.cisco.com, ns2.cisco.com, and ns3.cisco.com. Whenever a public domain is registered, it is mandatory to specify one or more authoritative name servers for that domain. These name servers are responsible for resolving that public domain to IP addresses.

Let's go through the steps of a DNS lookup from the perspective of a client that is trying to resolve a domain to an IP address (see Figure 18-3):

Step 1. The client query travels from the client machine to the configured DNS server on that machine. This DNS server is the DNS recursive resolver server.

Step 2. The DNS recursive resolver queries a DNS root name server.

Step 3. The root server responds to the recursive resolver with the TLD server for the requested last portion of the hostname. In the case of cisco.com, this would be the .com TLD server.

Step 4. The resolver queries the .com TLD server next.

Step 5. The TLD server responds with the IP address of the authoritative name server—in this example, the DNS server responsible for the cisco.com domain.

Step 6. The resolver sends the query to the authoritative name server.

Step 7. The authoritative name server responds with the IP address of the cisco.com web server.

Step 8. The DNS resolver responds to the client with the IP address obtained from the authoritative name server.

Step 9. The client can finally send the web page request to the IP address of the web server.

Step 10. The web server returns the web page to the client, and the browser renders it for the user.

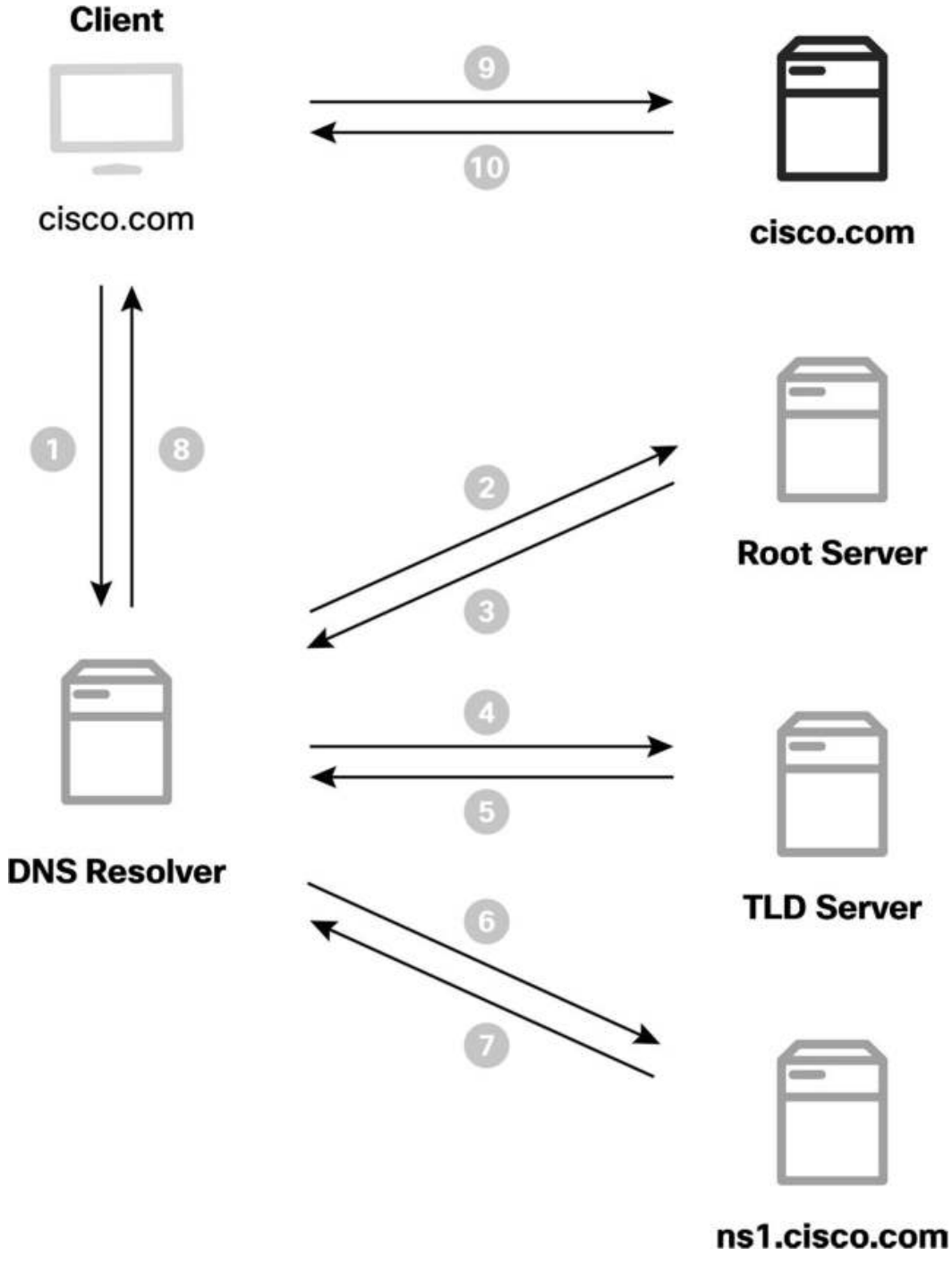


Figure 18-3 *DNS Name Resolution Steps*

A caching mechanism is available with DNS so that the client queries can be resolved as quickly as possible. DNS caching means temporarily storing results obtained during previous requests on DNS servers that are close to the client. Caching DNS resolution data makes it possible to resolve client queries earlier in the DNS lookup chain, which improves resolution time and reduces bandwidth and CPU consumption.

DNS uses both User Datagram Protocol (UDP) and Transmission Control Protocol (TCP) on port 53. General-purpose DNS implementations must support both; TCP is also also used for zone transfers and commonly for large or truncated responses. Several different types of records are stored in the DNS database, including IP addresses (A records for IPv4 and AAAA records for IPv6 addresses), SMTP mail exchangers (MX records), IP addresses of name servers (NS records), and alias records (CNAME). Although it was not intended to be used as a general-purpose database, DNS has been extended to store many types of additional information.

The Internet Engineering Task Force (IETF) has published several Requests for Comments (RFCs) related to DNS over the years. Some of the most important ones are RFC 1034: *Domain Names—Concepts and Facilities*, RFC 1035: *Domain Names—Implementation and Specification*, and RFC 1123: *Requirements for Internet Hosts—Application and Support*.

Network Address Translation (NAT)

When Internet Protocol (IP) was created, very few people, if any, were expecting it to support a global network of billions of interconnected devices. As discussed in earlier chapters, IPv4 addresses are 32 bits long, which means they can uniquely address a bit more than 4 billion endpoints. This number was fine and out of reach for a long time, but as the number of endpoints connecting to the Internet grew exponentially, it was clear that 4 billion addresses would not be enough to uniquely identify all the connected devices. At that point, work started for a new version of IP, IPv6, which defines 128-bit addresses and is able to uniquely identify trillions of endpoints. At the same time, it was clear that an overnight switchover from one IP version to another would be an impossible feat on the Internet, so several temporary solutions were proposed to ease the transition and extend the life of the IPv4-based Internet.

Network Address Translation (NAT) is one of the solutions to preserve the dwindling number of public IPv4 addresses. NAT reuses private IPv4 address blocks in internal networks and translates those addresses into public and unique IPv4 addresses at the borders of the internal networks. RFC 1918: *Address Allocation for Private Internets* declared a set of subnets private and unroutable on the global Internet. The subnets 10.0.0.0/8, 172.16.0.0/12, and 192.168.0.0/16 are extensively used in all private networks in the world, from enterprise networks to small office/home office networks. RFC 1918 is only one category of non-global IPv4 space. Other special-purpose ranges include loopback, link-local, shared address space, documentation ranges, multicast, and additional reserved blocks.

NAT is mostly used to translate between private RFC 1918 subnets and public IPv4 subnets. This translation happens at the exit points from the private networks, which in most cases are firewalls or border routers. NAT can also be used to translate between private networks and other private networks. In the case of mergers and acquisitions, it is possible that the enterprise that was acquired uses the same private IPv4 subnets as the acquiring company. To be able to exchange traffic between these networks that are addressed with the same IP addresses, NAT can be used to perform address translation. Basically, whenever an IP address needs to be translated into another address, NAT can be used.

Traditional NAT is described in Informational RFC 3022: *Traditional IP Network Address Translator (Traditional NAT)*, which obsoletes RFC 1631. A large number of Cisco and third-party routers and firewalls support NAT. The devices that perform IP address translations are usually situated and route data traffic between the internal network and the outside world or the public Internet. During the NAT configuration phase, an internal subnet or a set of subnets is defined as being internal, or "inside"; these are usually the private IP subnets used internally in the enterprise. As a second step in the configuration process, a single public IP address or an external, or "outside," pool of IP addresses are defined. With this information, the border device can perform IP address translation between the internal and external worlds. Several types of NAT are available:

- **Static NAT (static NAT):** Static NAT defines a one-to-one mapping between the internal IP address and its public IP address correspondent.
- **Dynamic NAT (dynamic NAT):** With dynamic NAT, the internal subnets that are permitted to have outside access are mapped to a pool of public IP addresses. The pool of public IP addresses is generally smaller than the sum of all the internal

subnets. This is usually the case in enterprise networks, where public IPv4 addresses are scarce and expensive, and a one-to-one mapping of internal to external subnets is not feasible. Reusing the pool of public IP addresses is possible because not all internal clients will access the outside world at the same time.

- *Port Address Translation (PAT* **or overloading):** PAT takes the dynamic NAT concept to the extreme and translates all the internal clients to one public IP address, using TCP and UDP ports to distinguish the data traffic generated by different clients. This concept is explained in more detail later in this chapter.

The type of NAT used in a particular situation depends on the number of public IP addresses defined and how the translation process is implemented.

To illustrate how NAT works, let's assume that a client connected to an enterprise network uses private IPv4 addressing. As the client generates data traffic and tries to connect to the Internet, the traffic makes its way to the enterprise border device. The border device looks up the destination of the traffic and the NAT configuration. If the client IP address is part of the internal subnets that have to be translated, it creates an entry in its NAT table with the source and destination IP addresses, it changes the source IP address of the packet from the internal private IP address to the public IP address, and it forwards the packet toward its destination. As the data traffic is received at the destination, the destination node is unaware that the traffic received went through the NAT process. The IP addresses in the response traffic are swapped, and as the data traffic is received by the border device, a lookup in the NAT table is done for the entry that was created as the traffic was exiting the network. The entry is matched, and the translation is done again—but this time in the reverse order, from the public IP address back to the private IP address of the client—and the traffic is routed back to the original source.

With PAT, the same type of table that keeps track of private to public translations and vice versa is created on the border device, but in this case TCP and UDP ports are also taken into account. For example, if the client generates web traffic and is trying to reach a web server on the Internet, the randomly generated TCP source port and the destination TCP port 443 for HTTPS are also included in the network translation table. In this way, many simultaneous translation entries can share one public IP address. Capacity depends on usable ports, destination tuples, protocol state, timeouts, and platform resources rather than a fixed number of clients. Figure 18-4 illustrates PAT, which is also called *overloading* because many internal private IP addresses are translated to only one public IP address.

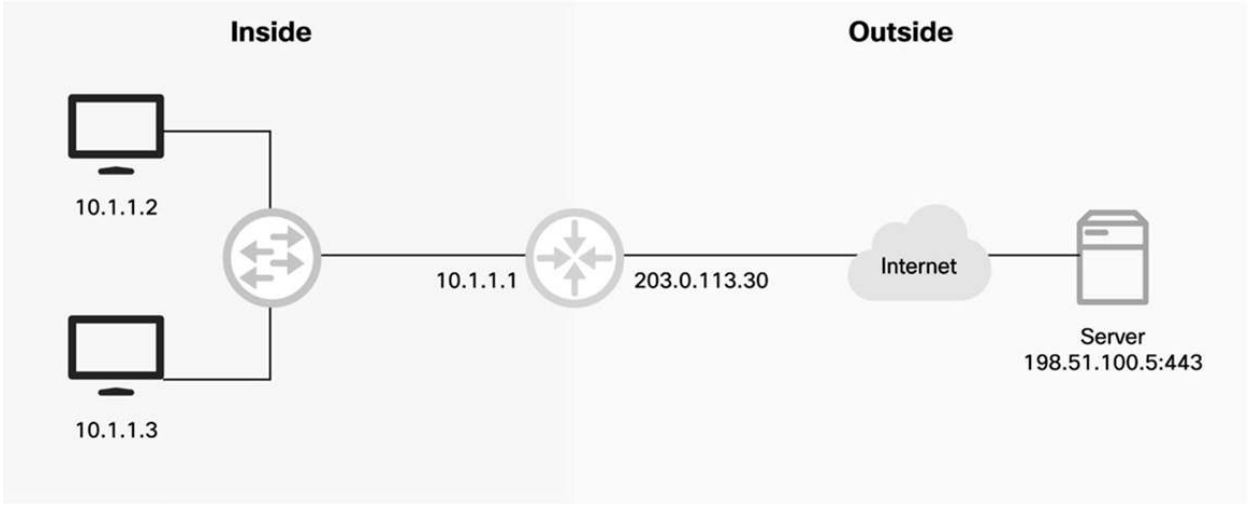


Figure 18-4 *Port Address Translation*

Some of the benefits of NAT are as follows:

- **Reduced costs for renting public IPv4 addresses:** With dynamic NAT and especially with PAT, a large number of private internal endpoints can be hidden behind a much smaller number of public IP addresses. Since public IPv4 addresses have become a rare commodity, there is a price for each public IPv4 address used. Large cost savings are possible by using a smaller number of public addresses.
- **Conserving public IPv4 address space:** The Internet would not have witnessed the exponential growth of the past few decades without NAT. Extensively reusing RFC 1918 private IP addresses for internal networks helped slow the depletion of IPv4 address space.
- **Reduced direct address visibility:** Hiding internal addresses can limit some reconnaissance, but NAT is not a security control or a substitute for firewall policy, host security, and access control.
- **Flexibility and cost savings when changing ISP connectivity:** In cases in which external network connectivity needs to be changed and migrations to new ISPs are required, configuration changes need to be performed only on the network border

devices, but the rest of the internal network does not need to be re-addressed. This results in massive cost savings, especially in large enterprise networks.

Although the advantages of NAT far outweigh the disadvantages in most cases, there are some disadvantages, including the following:

- **Loss of end-to-end functionality:** Some applications, especially for real-time voice and video signaling, are sensitive to changes in IP header addressing. While establishing these types of real-time voice and video sessions, headers exchanged at the application layer contain information pertaining to the private non-globally routable IP addresses of the endpoints. When using NAT in these cases, the IP addresses in the Layer 3 headers differ from the IP addresses contained in the application layer headers. This results in an inability to establish end-to-end voice and video calls.
- **Loss of end-to-end traceability and visibility:** Troubleshooting end-to-end connectivity issues is especially challenging in networks that use NAT.
- **Degradation of network performance:** NAT operations on border devices are usually not resource intensive, and several mechanisms have been implemented to make this impact even lower. Still, a border device needs to take an additional step before forwarding the data traffic toward its destination, and that means additional delay and consumption of memory and CPU resources.

NAT is extensively used in networks that use IPv4 addressing. One of the requirements for IPv6 was to restore end-to-end connectivity between all endpoints on the network, so NAT is not popular in IPv6 networks.

Simple Network Management Protocol (SNMP)

Simple Network Management Protocol (SNMP) is an application layer protocol used for monitoring and configuring devices. It was originally developed in the 1980s, and the IETF has published several RFCs covering SNMP since then. As networks were becoming larger and more complicated in those days, a need to remotely monitor and manage devices arose. Following are several versions of SNMP that have been released through the years:

- SNMP version 1 (SNMPv1)
- SNMP version 2 (SNMPv2)
- SNMP version 2c (SNMPv2c)
- SNMP version 3 (SNMPv3)

While versions 1 and 2 of SNMP are rarely used anymore, versions 2c and 3 are used extensively in production environments. SNMPv2 adds 64-bit counter support and includes additional protocol operations. With SNMPv3, the focus is on security, so additional features like authentication, encryption, and message integrity were added to the protocol specification.

The following are some of the advantages of SNMP:

- It provides a single framework for monitoring many different kinds of devices.
- It is based on open standards documented in IETF RFCs.
- It is easily extensible.

There are also disadvantages with SNMP, including the following:

- The lack of writable Management Information Bases (MIBs) leads to poor configuration capabilities. SNMP is rarely used for configuration purposes.
- The lack of atomic transactions makes rollbacks to previous states difficult.
- SNMP is slow for monitoring purposes when large amounts of operational data need to be retrieved.
- It is CPU and memory resource intensive when large amounts of performance metrics are retrieved.

Even though SNMP was originally designed to support both monitoring and configuration capabilities, historically only the monitoring capabilities were used. The SNMP specification defines the following three components:

- Managed devices
- SNMP agent
- SNMP manager

Managed devices are those devices that are monitored and managed through SNMP. They implement an SNMP interface through which the SNMP manager monitors and controls the device. The SNMP agent is the software component that runs on the managed device and translates between the local management information on the device and the SNMP version of that information. The SNMP manager, also called the *Network Management Station (NMS)*, is the application that monitors and

controls the managed devices through the SNMP agent. The SNMP manager offers a monitoring and management interface to network and system administrators. The SNMP components and their interactions are illustrated in Figure 18-5.

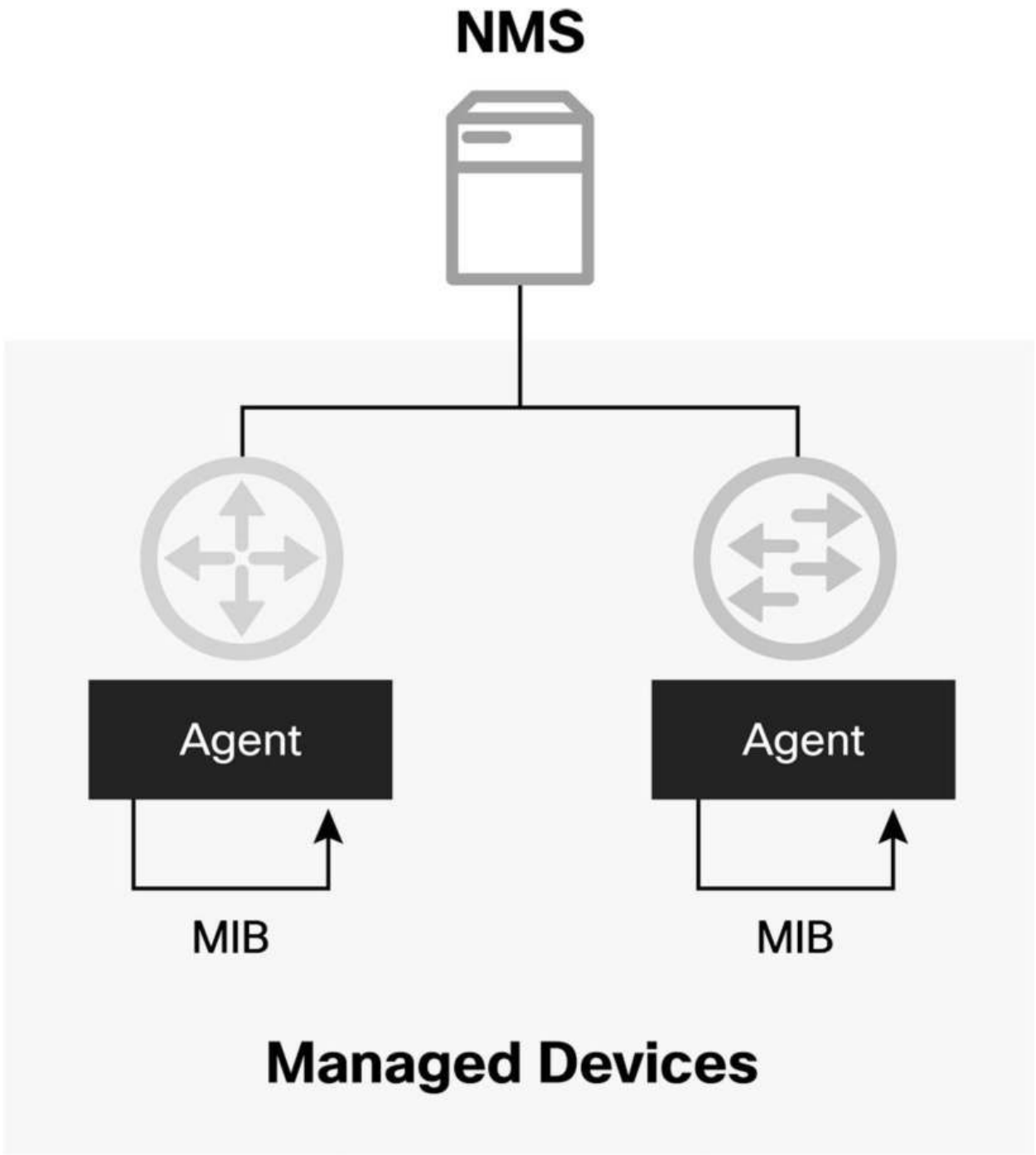


Figure 18-5 *SNMP Components*

The SNMP agent listens on UDP port 161 for requests from the SNMP manager. SNMP also supports notifications, which are SNMP messages that are generated on the managed device when significant events take place. Through notifications, the SNMP agent notifies the SNMP manager about these critical events. The NMS listens for these notifications on UDP port 162. SNMP data structures that facilitate the exchange of information between the SNMP agent and the NMS are organized as a list of data objects called a *Management Information Base (MIB)*. You can think of a MIB as a map of all components of a device that are managed by SNMP. An NMS can query supported numeric OIDs without loading a MIB. Loading the relevant MIB modules provides symbolic names, types, descriptions, and decoding that make management data easier to use.

A MIB is organized as a tree-like structure with unique variables represented as leaves. Each variable in the tree is uniquely identified by an *Object Identifier (OID)*. Operational data such as CPU temperature, fan speed, and outbound packets on an interface are all represented through OID values. For the NMS to obtain any device's operational metric, it needs to send an

SNMP GET packet that includes OID values for each metric of interest. The SNMP agent receives the packet and looks up the OIDs in the MIB, and if the device implements the requested information, it returns the information to the NMS.

Several types of SNMP messages are used to communicate between the SNMP manager and the agent:

- **GetRequest:** The NMS uses this message to request the value of a variable or list of variables. The agent returns the request information in a Response message.
- **SetRequest:** The SNMP manager uses this message to request a change to a variable or a list of variables. The agent returns a Response message containing the new values for the variables.
- **GetNextRequest:** The SNMP manager uses this message to discover available variables and their values. This is a common operation used to "walk" the entire MIB of an agent. The agent returns a Response message that contains the requested information.
- **GetBulkRequest:** This optimization of the GetNextRequest message, which was introduced with SNMPv2, contains multiple iterations of the GetNextRequest call.
- **Response:** The SNMP agent generates this message, which contains the information the SNMP manager requested.
- **Trap:** The SNMP agent generates this notification message to signal to the SNMP manager when critical events take place on the managed device.
- **InformRequest:** The SNMP agent sends this acknowledged notification to the SNMP manager. Keep in mind that SNMP runs over UDP, which is a connectionless protocol, and packets might get lost during transmission. A rudimentary acknowledgment mechanism is implemented through InformRequest messages.

SNMPv1 and SNMPv2c community names act as shared credentials between managers and agents. Requests are accepted only when the community string and other access controls match. Access can be read-only or read-write and limited by MIB views and source-address controls. You should use SNMPv3 authentication and privacy where available.

Network Time Protocol (NTP)

Accurately keeping track of time is critical in today's IT infrastructure. As IT infrastructure becomes more and more ingrained in business processes and success, every second of downtime when the infrastructure is not available translates into loss of revenue for the business. In extreme cases, this downtime can lead to loss of customers and bankruptcy. Service-level agreements (SLAs) are contracts between providers and consumers of infrastructure. As an example, stringent SLA contracts require 99.999% uptime, which in the case of Internet service providers translates into no more than 5 minutes of downtime per year. It is critically important to make sure that a consistent, uniform, and correct view of time exists on all the devices in a network. Time is fundamental when measuring SLAs and enforcing contracts. Inaccurate time can lead to service disruptions. As a simple example, with web traffic, HTTPS TLS connections will not even be established if the time on the web server or the client is not accurate.

The system clock on each device is the heart of the time service. The system clock runs from the second the operating system starts; it keeps track of the date and time. The system clock can be set to update from different time sources and can be used to distribute time to other devices. *Network Time Protocol (NTP)* enables a device to update its clock from a trusted network time source and serve time to other devices, enabling groups of devices to be time synchronized. Most devices contain battery-powered clocks that keep track of date and time across restarts and power outages.

The main role of NTP is to synchronize the time on a network of devices. This protocol was developed by the IETF, and the latest version, version 4, is defined in RFC 5905: *Network Time Protocol Version 4: Protocol and Algorithms Specification*. NTP uses UDP at the transport layer, and port 123 is reserved for it. NTP works based on a client/server architecture, with NTP servers providing the time to clients. Authoritative time sources are servers that have attached radio clocks or atomic clocks, making them extremely accurate. NTP has the role of distributing time to all the devices connected to the network. Multiple NTP servers can coexist at the same time on the same subnet, and clients can use all of them for time synchronization. NTP clients poll the time servers at intervals managed dynamically by conditions on the network such as latency and jitter. NTP adjusts poll intervals dynamically based on protocol state and clock and network behavior; no single fixed interval is universally sufficient.

The concept of strata is used in NTP to describe how many hops or devices away a client is from an authoritative time source. NTP servers that are most authoritative and are directly connected to very accurate time sources are in stratum 1. A stratum 2 time server receives time from a stratum 1 server, and so on. When a client receives time from multiple servers, NTP selection, clustering, and combining algorithms evaluate offset, delay, dispersion, jitter, reachability, and falsetickers. Stratum is one input to topology and distance, not a simple trust ranking that always selects the lowest value.

NTPv4 defines client/server, symmetric active/passive, and broadcast communication variants. Configuration and discovery are separate from protocol mode. You can manually configure NTP clients to establish a connection and to associate and solicit time updates from NTP servers by simply statically configuring the hostname or the IP addresses of the servers. In local-area networks within the same subnet, NTP can be configured to use broadcast messages instead. Configuration in this situation is simpler, because each device can be configured to either send or receive broadcast messages. With broadcast NTP messages, a slight loss of accuracy occurs, since the flow of information is only one way.

Because time accuracy is critical in today's infrastructure, it is recommended to implement all security features that come with NTP. Two NTP security features are most commonly used:

- Use an appropriate authentication mechanism between clients and servers. For modern client/server NTP, Network Time Security (NTS) uses TLS for authenticated key establishment and AEAD to protect NTP extension fields; basic time data is not fully encrypted.
- NTP associations should be limited to only trusted servers through access control lists.

When accurate time matters, current IETF best practice recommends at least four independent, diverse upstream time sources. You should use public NTP services only under their stated policies.

Layer 2 Versus Layer 3 Network Diagrams

Network diagrams are a critical component in the documentation process for any network. Just as software developers document their code for easier maintenance and sharing, network engineers document networks by building network diagrams. Having accurate network diagrams is very important, especially when troubleshooting network issues. The network has become a business differentiator for all companies, and each second of downtime costs money. Having up-to-date network diagrams makes it much easier and faster to troubleshoot network issues and decrease the mean time to resolution (MTTR) when problems arise.

Several different types of network diagrams can be created, depending on what needs to be emphasized and documented. Layer 1 network diagrams can be used to show physical connections, including how network devices are connected and what type of cables (twisted pair, fiber optics, and so on) are being used. These diagrams can also show patch panel port connectivity and availability and everything else that is in the confines of physical connectivity.

Layer 2 network diagrams contain information related to all Layer 2 devices, protocols, and connectivity in a network. Such a diagram shows all the switches in the network and how they are interconnected, including how the ports on one switch connect to the ports on another switch, which VLANs are configured on each switch, which ports are configured as trunks and what VLANs are allowed on them, which ports are configured as port channels, and any other Layer 2 information. Depending on the size of the network, a Layer 2 diagram can be split into multiple documents.

A Layer 3 network diagram captures all the network information available at Layer 3. Such a diagram should show all devices that operate at Layer 3—routers, firewalls, load balancers, and so on—and how they are interconnected. Information about IP addressing and subnets, routing protocols, first-hop redundancy protocols (such as HSRP, VRRP, and GLBP), Layer 3 port channels, and Internet connectivity should also be included at a minimum. More information can also be included in these diagrams, but care should be taken to avoid including too much information or dynamic information that is bound to change often.

Several software tools are available for creating network diagrams. You can create network diagrams manually by using tools such as Microsoft Visio and draw.io, or you can have them created automatically with controller-based solutions such as Cisco Catalyst Center (formerly Cisco DNA Center). With Cisco Catalyst Center, the controller automatically discovers all the devices in the network and how they are interconnected, and it uses this information to build network diagrams, topologies, and databases.

It is a good idea to store network diagrams in version control systems such as Git. Networks are dynamic and evolve over time, and it is important to capture this evolution in the diagrams. Git offers a proven solution to storing and keeping track of changes in documents, so it is ideal for storing network diagram versions. With infrastructure as code and solutions such as Cisco Modeling Labs (CML), network diagrams and topologies are stored as YAML files and can be used as part of CI/CD infrastructure configuration automation pipelines.

Troubleshooting Application Connectivity Issues

Applications are becoming more and more complicated, with many moving components that need to communicate with each other over the network. With the advent of the microservices architecture for applications and the ubiquity of APIs, the network is critical to the functionality of applications. It is important to know how to troubleshoot connectivity issues and how they might affect the performance of applications.

Network connectivity may not function as expected for a variety of reasons. Whenever an application or a website or any target destination that is being accessed over a network is not behaving as expected, the network takes the blame. We will see next that while the network might be the culprit in some instances, applications stop responding as expected for many other reasons.

Network troubleshooting usually follows the OSI layers, discussed in Chapter 16, "Network Fundamentals." It can occur from top to bottom, beginning at the application layer and going down the layers all the way to the physical layer, or it can occur from bottom to top. This section looks at a typical bottom-to-top troubleshooting session that starts from the physical layer and goes up the stack toward the application layer. The troubleshooting steps discussed here can also be followed in the reverse order, if desired.

First and foremost, it is important to know how an endpoint device connects to the network at the physical layer: with a wired or wireless connection. If the connection to the network is wired through an Ethernet cable, the network interface card should come online, and electrical signals should be exchanged with the switch port to which the NIC is connected. Depending on the operating system of the client connecting to the network, the status of the connection will show as solid green or will display as "enabled" or "connected" in the network settings. If the NIC status shows as connected, the physical layer is working as expected, and the troubleshooting process can proceed to the next layer. If the NIC doesn't show as connected or enabled, there could be several causes, including the following:

- Misconfigured or disabled switch port
- Defective network cable
- Defective wall network port
- Incorrect cabling in the patch panel
- Defective network interface card

Troubleshooting at the physical layer revolves around making sure an uninterrupted physical connection exists between the client and the switch port to which it connects.

If the connection to the network is wireless, it is important to ensure that the wireless network interface card is turned on and that it can send and receive wireless signals to and from the nearest wireless access point. Being within the service range of a wireless access point is also important, and usually the closer the client is to the access point, the better network performance it should experience.

Troubleshooting at the data link layer, or Layer 2, means confirming that the endpoint and switch learn the expected link-layer information. On a host or Layer 3 device, the **arp** command shows the ARP or neighbor cache, which maps IP addresses to MAC addresses. On a switch, the **show mac address-table** command displays MAC-address and VLAN-to-port mappings. Relevant entries in these distinct tables can support the diagnosis, but a Layer 2 switch does not need an ARP entry for every forwarded host. Some of the issues that cause the data link layer not to function as expected are as follows:

- Misconfigured switch port
- Layer 2 access control lists
- Misconfigured Spanning Tree Protocol
- Missing or misconfigured VLANs

At Layer 3 (the Internet layer), IP connectivity and reachability have to work. The network client should be configured with the correct IP address, subnet mask, and default gateway. This is usually done using DHCP servers, and in rare cases it may be manually configured. Built-in operating system tools such as **ifconfig** and **ping** can be used at this layer to help with troubleshooting. **ifconfig** (or **ipconfig** on Microsoft Windows) is a network utility that retrieves and can modify the configuration and status of all network interfaces present on the client endpoint. **ping** is another popular network tool that tests ICMP echo

reachability to an endpoint. If the destination of the client data traffic is in a different subnet than the client is connected to, that means that the traffic has to be routed through the default gateway of the client subnet. Layer 3 connectivity between data traffic source and destination can be checked using the **ping** command. A successful ping confirms ICMP reachability at that moment. A failed ping is inconclusive because ICMP can be filtered or rate-limited even when application traffic works, so test the relevant route and application protocol as well. If reachability is not established, look for problems that can cause connectivity issues at Layer 3, including these:

- Misconfigured IP information (IP address, subnet mask, default gateway) on the client device
- Layer 3 access control lists that blocks data traffic
- Routing protocol issues causing black-holing or incorrect routing of traffic

Troubleshooting at the transport layer means making sure that the network clients can access the TCP or UDP ports on which the destination applications are running. For example, in the case of web traffic, it is important to verify that the client can connect to TCP ports 80 (HTTP) and/or 443 (HTTPS) on the web server. In some cases, web servers are configured to listen on esoteric ports such as 8080, so it is important to know the correct port on which the destination application is running. Networking tools such as **curl** and custom **telnet** commands specifying the application port can be used to ensure that transport layer connectivity can be established end to end between the source and destination. If a transport layer connection cannot be established, you need to look for issues such as these:

- Firewall access control lists blocking data traffic based on TCP and UDP ports
- Misconfigured applications and listening ports
- Misconfigured load balancers
- Presence of proxy servers that are intercepting the traffic and denying connectivity
- Misconfigured PAT

Other common problems that affect application connectivity are DNS related. As discussed earlier in this chapter, DNS plays a critical role in resolving domain names to IP addresses. If DNS is malfunctioning for some reason, end-to-end connectivity is impacted. Network tools such as **nslookup** can be used to troubleshoot DNS functionality. The following problems commonly cause DNS issues:

- Misconfigured DNS resolver on the network client
- Wrong hostname specified
- Invalid DNS server configuration
- Missing or incorrect DNS entry

Even if end-to-end connectivity at the transport layer is verified, there can still be issues on the network that cause connections to applications to fail. These issues are usually rather difficult to discover and troubleshoot and are related to data traffic load and network delay. The difficulty with these issues comes from the fact that they are difficult to reproduce and can be temporary in nature, caused by short spikes in network traffic. Networking tools such as **iperf** can be used to dynamically generate traffic and perform load stress on the network to ensure that large amounts of data can be transported between the source and destination. Implementing quality of service (QoS) throughout the network can help with these problems. With QoS, traffic is categorized in different buckets, and each bucket gets separate network treatment. For example, you can classify traffic such as voice and video as real-time traffic by changing QoS header fields in the Layer 2 data frames and Layer 3 packets so that when switches and routers process this type of traffic, they give it a higher priority and guaranteed bandwidth, if necessary.

At the application layer, network tools such as **tcpdump** can be used to capture actual data traffic received on any of the network interfaces of either the source device or the destination device. Comparing between sent and received data can help in troubleshooting connectivity issues and determining the root cause of a problem. Slow or no responses at the application layer could indicate an overloaded backend database, misconfigured load balancer, or faulty code introduced through new application features.

Exam Preparation Tasks

As mentioned in the section “How to Use This Book” in the Introduction, you have a couple of choices for exam preparation: the exercises here, Chapter 19, “Final Preparation,” and the exam simulation questions on the companion website.

Review All Key Topics

Review the most important topics in this chapter, noted with the Key Topic icon in the outer margin of the page. Table 18-2 lists these key topics and the page number on which each is found.

Table 18-2 *Key Topics*

Key Topic Element	Description	Page Number
List	Benefits of DHCP	
List	Four phases of DHCP operations	
List	DNS resolution process components	
Paragraph	User Datagram Protocol (UDP)	
Paragraph	Network Address Translation (NAT)	
Paragraph	Traditional NAT, described in Informational RFC 3022	
Paragraph	Port Address Translation (PAT)	
List	Advantages of SNMP	
Paragraph	SNMP agent listening for requests from the SNMP manager	
List	SNMP message types	
Paragraph	Main role of NTP	
Paragraph	Two forms of communication between NTP clients and servers	
Paragraph	Layer 2 network diagrams	
Paragraph	Layer 3 network diagrams	
Paragraph	Network troubleshooting	
Paragraph	Troubleshooting at the data link layer	
Paragraph	Troubleshooting at the Internet layer	
Paragraph	Troubleshooting at the transport layer	
Paragraph	Network issues with application connection failures	

Define Key Terms

Define the following key terms from this chapter and check your answers in the glossary:

Dynamic Host Configuration Protocol (DHCP)
Domain Name System (DNS)

top-level domain (TLD) name server
Network Address Translation (NAT)
Port Address Translation (PAT)
Simple Network Management Protocol (SNMP)
Network Management Station (NMS)
Management Information Base (MIB)
Object Identifier (OID)
Network Time Protocol (NTP)

Chapter 19. Final Preparation

The first 18 chapters of this book cover the technologies, protocols, design concepts, and considerations required to be prepared to pass the 200-901 CCNA Automation CCNAAUTO exam. While those chapters supply the detailed information, most people need more preparation than simply reading the first 18 chapters of this book. This chapter provides a set of tools and a study plan to help you complete your preparation for the exam.

This short chapter has three main sections. The first section helps you get ready to take the exam, and the second section lists the exam preparation tools useful at this point in the study process. The third section provides a suggested study plan you can follow, now that you have completed all the earlier chapters in this book.

Getting Ready

Here are some important tips to keep in mind to ensure that you are ready for this rewarding exam:

- **Build and use a study tracker:** Consider using the exam objectives shown in this chapter to build a study tracker for yourself. Such a tracker can help ensure that you have not missed anything and that you are confident for your exam. As a matter of fact, this book offers a sample Study Planner as a website supplement.
- **Think about your time budget for questions on the exam:** When you do the math, you will see that, on average, you have one minute per question. While this does not sound like a lot of time, keep in mind that many of the questions will be very straightforward, and you will take 15 to 30 seconds on those. This leaves you extra time for other questions on the exam.
- **Watch the clock:** Check in on the time remaining periodically as you are taking the exam. You might even find that you can slow down pretty dramatically if you have built up a nice block of extra time.
- **Get some earplugs:** The testing center might provide earplugs but get some just in case and bring them along. There might be other test takers in the center with you, and you do not want to be distracted by their screams. I personally have no issue blocking out the sounds around me, so I never worry about this, but I know it is an issue for some.
- **Plan your travel time:** Give yourself extra time to find the center and get checked in. Be sure to arrive early. As you test more at a particular center, you can certainly start cutting it closer time-wise.
- **Get rest:** Most students report that getting plenty of rest the night before the exam boosts their success. All-night cram sessions are not typically successful.
- **Bring in valuables but get ready to lock them up:** The testing center will take your phone, your smartwatch, your wallet, and other such items and will provide a secure place for them.
- **Take notes:** You will be given note-taking implements and should not be afraid to use them. I always jot down any questions I struggle with on the exam. I then memorize them at the end of the test by reading my notes over and over again. I always make sure I have a pen and paper in the car, and I write down the issues in my car just after the exam. When I get home—with a pass or fail—I research those items!

Tools for Final Preparation

This section lists some information about the available tools and how to access the tools.

Pearson Cert Practice Test Engine and Questions on the Website

Register this book to get access to the Pearson IT Certification test engine (software that displays and grades a set of exam-realistic multiple-choice questions). Using the Pearson Cert Practice Test Engine, you can either study by going through the questions in Study mode or take a simulated (timed) CCNA Automation CCNAAUTO exam.

The Pearson Test Prep practice test software comes with two full practice exams. These practice tests are available to you either online or as an offline Windows application. To access the practice exams that were developed with this book, please see the instructions in the card inserted in the sleeve in the back of the book. This card includes a unique access code that enables you to activate your exams in the Pearson Test Prep software.

Accessing the Pearson Test Prep Software Online

The online version of this software can be used on any device with a browser and connectivity to the Internet, including desktop machines, tablets, and smartphones. To start using your practice exams online, simply follow these steps:

Step 1. Go to http://www.PearsonTestPrep.com.

Step 2. Select **Pearson IT Certification** as your product group.

Step 3. Enter your email and password for your account. If you don't have an account on PearsonITCertification.com or CiscoPress.com, you need to establish one by going to PearsonITCertification.com/join.

Step 4. In the **My Products** tab, click the **Activate New Product** button.

Step 5. Enter the access code printed on the insert card in the back of your book to activate your product.

Step 6. The product will now be listed in your My Products page. Click the Exams button to launch the exam settings screen and start your exam.

Accessing the Pearson Test Prep Software Offline

If you wish to study offline, you can download and install the Windows version of the Pearson Test Prep software. You can find a download link for this software on the book's companion website, or you can just enter this link in your browser:

http://www.pearsonitcertification.com/content/downloads/pcpt/engine.zip

To access the book's companion website and the software, simply follow these steps:

Step 1. Register your book by going to PearsonITCertification.com/register and entering the ISBN **9780135368084**.

Step 2. Respond to the challenge questions.

Step 3. Go to your account page and select the Registered Products tab.

Step 4. Click on the Access Bonus Content link under the product listing.

Step 5. Click the Install Pearson Test Prep Desktop Version link in the Practice Exams section of the page to download the software.

Step 6. When the software finishes downloading, unzip all the files onto your computer.

Step 7. Double-click the application file to start the installation and follow the onscreen instructions to complete the registration.

Step 8. When the installation is complete, launch the application and click the Activate Exam button on the My Products tab.

Step 9. Click the Activate a Product button in the Activate Product Wizard.

Step 10. Enter the unique access code from the card in the sleeve in the back of your book and click the Activate button.

Step 11. Click Next and then click the Finish button to download the exam data to your application.

Step 12. You can now start using the practice exams by selecting the product and clicking the Open Exam button to open the exam settings screen.

Note that the offline and online versions sync together, so saved exams and grade results recorded on one version will be available to you in the other version as well.

Customizing Your Exams

Once you are in the exam settings screen, you can choose to take exams in one of three modes:

- Study mode
- Practice Exam mode
- Flash Card mode

Study mode allows you to fully customize your exam and review answers as you are taking the exam. This is typically the mode you use first to assess your knowledge and identify information gaps. Practice Exam mode locks certain customization options in order to present a realistic exam experience. Use this mode when you are preparing to test your exam readiness. Flash Card mode strips out the answers and presents you with only the question stem. This mode is great for late-stage preparation, when you really want to challenge yourself to provide answers without the benefit of seeing multiple-choice options. This mode will not provide the detailed score reports that the other two modes provide, so you should not use it if you are trying to identify knowledge gaps.

In addition to these three modes, you will be able to select the source of your questions. You can choose to take exams that cover all of the chapters, or you can narrow your selection to just a single chapter or the chapters that make up specific parts in the book. All chapters are selected by default. If you want to narrow your focus to individual chapters, simply deselect all the chapters and then select only those on which you wish to focus in the Objectives area.

You can also select the exam banks on which to focus. Each exam bank comes complete with a full exam of questions that cover topics in every chapter. Two exams are available to you online as part of the print book, as well as two additional exams of unique

questions. You can have the test engine serve up exams from all four banks or just from one individual bank by selecting the desired banks in the exam bank area.

There are several other customizations you can make to your exam from the exam settings screen, such as the time allowed for taking the exam, the number of questions served up, whether to randomize questions and answers, whether to show the number of correct answers for multiple-answer questions, and whether to serve up only specific types of questions. You can also create custom test banks by selecting only questions that you have marked or questions on which you have added notes.

Updating Your Exams

If you are using the online version of the Pearson Test Prep software, you should always have access to the latest version of the software as well as the exam data. If you are using the Windows desktop version, every time you launch the software, it will check to see if there are any updates to your exam data and automatically download any changes made since the last time you used the software. This requires that you are connected to the Internet at the time you launch the software.

Sometimes, due to a number of factors, the exam data might not fully download when you activate your exam. If you find that figures or exhibits are missing, you might need to manually update your exams.

To update a particular exam you have already activated and downloaded, simply select the **Tools** tab and select the **Update Products** button. Again, this is only an issue with the desktop Windows application.

If you wish to check for updates to the Windows desktop version of the Pearson Test Prep exam engine software, simply select the **Tools** tab and click the **Update Application** button. This will ensure you are running the latest version of the software engine.

Premium Edition

In addition to the free practice exam provided on the website, you can purchase additional exams with expanded functionality directly from Pearson IT Certification. The Premium Edition of this title contains an additional two full practice exams and an eBook (in both PDF and ePub format). In addition, the Premium Edition title also has remediation for each question to the specific part of the eBook that relates to that question.

Because you have purchased the print version of this title, you can purchase the Premium Edition at a deep discount. There is a coupon code in the book sleeve that contains a one-time-use code and instructions for where you can purchase the Premium Edition.

To view the Premium Edition product page, go to www.informit.com/title/9780135368121.

Chapter-Ending Review Tools

Chapters 2 through 18 include several features in the “Exam Preparation Tasks” section at the end of the chapter. You might have already worked through these in each chapter. It can also be useful to use these tools again as you make your final preparations for the exam.

Suggested Plan for Final Review/Study

This section lists a suggested study plan from the point at which you finish reading through Chapter 18 until you take the 200-901 CCNA Automation CCNAAUTO exam. You can ignore this plan, use it as is, or just take suggestions from it.

The plan involves two steps:

Step 1. **Review key topics and “Do I Know This Already?” (DIKTA?) questions:** You can use the table that lists the key topics in each chapter or just flip the pages, looking for key topics. Also, reviewing the DIKTA? questions from the beginning of the chapter can be helpful for review.

Step 2. **Use the Pearson Cert Practice Test engine to practice:** The Pearson Cert Practice Test engine allows you to study using a bank of unique exam-realistic questions available only with this book.

Summary

The tools and suggestions listed in this chapter have been designed with one goal in mind: to help you develop the skills required to pass the 200-901 CCNA Automation CCNAAUTO exam. This book has been developed from the beginning to not just tell you the facts but to also help you learn how to apply the facts. No matter what your experience level leading up to when you take the exam, it is our hope that the broad range of preparation tools, and even the structure of the book, will help you pass the exam with ease. We hope you do well on the exam.

Appendix A. Answers to the “Do I Know This Already?” Quiz Questions

Chapter 1

1. A, B, D. The five levels of Cisco accreditation are Entry, Associate, Professional, Expert, and Architect.
2. A, B, E. Highlighting skills to employers and peers, increasing confidence, improving credibility, providing value to employers, providing a baseline of understanding, career advancement, and increased salary are some of the most common reasons candidates want to get certified.
3. A, E. Only two types of exams are necessary to obtain the CCNP Automation certification: the Technology Core exam and a single concentration exam.
4. A. A single exam is all that is required for the new CCNA certification.
5. A, B, C, E. Cisco Automation Platform doesn’t exist. The DevNet Automation Exchange is a place on DevNet to download fully tested and working use case–driven code-based examples.

Chapter 2

1. B. Waterfall is a type of SDLC.
2. B. Agile is an implementation of the Lean management philosophy for software development.
3. A, B, D. Model-View-Controller is a software design pattern used to create interactive graphical web-centric applications. It can be found in Django (a web app framework) as well as many web applications that require support for many types of clients.
4. A, C. The Observer pattern follows the publisher/subscriber model, where a client subscribes to the publisher and synchronizes its configuration state. This is perfect for one-to-many device configuration and management of event handling in infrastructure components.
5. C. BASH stands for Bourne Again Shell.
6. B. env | more pipes the contents of your environment variables to the **more** command and allows for page breaks.
7. B. Software version control is also commonly known as source code management.
8. D. Linus Torvalds, the father of Linux, created Git.
9. B. The three main structures tracked by Git are local workspace, index, and local repository.
10. D. To add a specific file to the Git index, you use the command **git add** followed by the name of the file you want to add.
11. A. One of the main benefits of conducting formal code reviews is to help you the developer, create higher-quality software.

Chapter 3

1. D. The correct command is **python3 -m (for module) venv myvenv** (which can be whatever you choose to name your virtual environment).
2. B, C. PyPI is a repository that holds thousands of Python modules that you can import. To install it, you can use python3 -m (module) pip install and the name of the package you want. You can also directly install it with the pip command.
3. B. PEP 8 is the style guide for Python syntax, and it specifies four spaces for each block of code. Tabs will work, and your editor may actually convert them automatically for you, but it is a good practice to follow the standard.
4. B. Comments are specified by the # or three single quotes ‘’’. The benefit of using the single quotes is that you can write multiline text.
5. A, B. Lists and dictionaries are both mutable, or changeable, data types. Integers and tuples must be replaced and can’t be edited, which makes them immutable.

6. B, D. You can create an empty dictionary object by assigning the function **dict()** to a Python object (in this example, **a**). You can insert dictionary values as well by using braces, {}, and key:value pairs that you assign to an object.
7. C. The **input()** function by default creates a string data type.
8. D. The only one that is valid as is would be **print('"hello world"')**. **print(hello, world)** would be valid only if there were two variables named **hello** and **world**; otherwise, it would produce an error. Since you don't know in this case whether these variables exist, **print('"hello world"')** is the correct answer.
9. D. All of the statements are correct regarding how an **If** statement operates in Python.
10. A, D. The **range()** function in Python allows you to set a range that is up to but not including the number specified. If you want to increment to 10, for example, you need to provide the number 11. A range can also count up or down based on the sign of the number provided.

Chapter 4

1. C. A function in Python uses the **def** keyword followed by a function name and parentheses with an optional argument inside.
2. D. Python has specific rules for variable names. A variable cannot be named using reserved keywords, such **True**, and must start with a letter or an underscore but not a number.
3. B. A docstring is text that describes the purpose and use of a Python function, and it is located on the very next line after the function definition. The docstring can be viewed with the **help()** function in Python.
4. B. Keyword arguments pass values by parameter name instead of by position, so sub(arg2=15, arg1=10) assigns 15 to arg2 and 10 to arg1 no matter what order they appear in. The other options pass values positionally or unpack a list.
5. A, B. The key components of object-oriented programming in Python are functions that can be performed on a data structure and attributes that are stored in an object.
6. A, B, C. The benefits of OOP are reusable code, easy-to-follow code, and low coupling/high cohesion between application components.
7. A, B. A Python class can be defined using the keyword class, a name for the class, and optionally a parent class to inherit attributes and functions from.
8. C. A method is a function that you can perform on an object. Methods are often defined in classes. They can also make use of externally defined functions as well.
9. B. Inheritance allows you to define a base class and then define another class using the previous class as a parent. The subsequent class includes all the capabilities of the parent class and can add to or override any attribute or method that needs to be different.
10. D. The **os** module allows you to interact with the file system and directory structure of the operating system.
11. A. Cisco built **pyATS** and released it to open source for testing Cisco infrastructure software.
12. A. Type hints document the argument and return types a function expects. Editors and type checkers use them to catch mistakes before the code runs. Python ignores them at runtime.

Chapter 5

1. C. The end-of-line is the last character of a line of text before the text wraps to the next line. This is identified as \n or as EoF (end of file).
2. A. In order to open a file for writing, you need to use the **open()** function and assign it to a file handler variable—in this case, data. In addition, you need to pass the name of the file and tell Python that you want to allow write access to it (using "w").
3. C. Using the **with open** combo to the text.csv file, you map it to the filehandle object. Map **csv_writer**, which is just another Python object, to the **csv.writer(filehandle)** function. Next, you write your data to the CSV file by calling the **.writerow** method.
4. B. The xmltodict module reads XML data.
5. A. To load a native JSON file into a Python string object, you use **loads()**, which stands for load string, and to convert a Python string into native JSON, you use **dump()**.
6. B. YAML stands for YAML Ain't Markup Language.
7. A. Error handling in Python can be conducted by using a **try-except-else-finally** block.

8. C. **finally** is executed every time the code runs through the **try** block. It is often used to clean up variables or alert the user to a success or failure event.
9. C. Test-driven development focuses on writing code that allows a previously written test (that naturally fails since no code was written) to succeed.
10. A, D. An integration test is for API verification, and a function test verifies that your application meets the agreed-upon requirements of how it should operate.
11. B. unittest.TestCase is a special class that is used to access the unittest module's capabilities.

Chapter 6

1. A. Southbound APIs send information down to devices within the network.
2. A, B. Because asynchronous APIs do not have to wait for replies, they reduce the time required to process data.
3. A, D, E. SOURCE and PURGE do not exist. GET, POST, PUT, PATCH, and DELETE are the HTTP functions.
4. A. Both API keys and custom tokens are commonly used within API authentication.
5. C. SOAP stands for Simple Object Access Protocol.
6. A, B, D, E. The four main components of a SOAP message are the envelope, header, body, and fault. The fault is an optional component.
7. A. RPCs are blocked during the waiting periods. Once a procedure is executed and the response is sent from the server and received on the client, the execution of the procedure continues. This is similar to a synchronous API.

Chapter 7

1. A, B. In order to make a successful call—whether it is a GET or a POST—a client must have the URL, the method, an optional header, and an optional body.
2. C. TRIGGER is not a defined HTTP method.
3. B, D. Webhooks are like callback functions for the web; they handle notifications from the server. Also, these notifications are triggered by events.
4. B. The 3xxx HTTP codes are for redirection. When a resource is moved, a server sends a 3xx code.
5. B. Sequence diagrams model the interactions between various objects in a single use case.
6. C. Code on demand provides flexibility to a client by allowing it to download code. For example, a client can download JavaScript and execute it.
7. D. Rate-limiting techniques help in limiting the security surface, allowing various business models (from freemium to paid), and improving the efficiency of the entire system.
8. B. The processes of encoding and decoding JSON are usually called serialization and deserialization, respectively. Python dictionaries (dicts) are used as standard data types for a lot of Python request functions.

Chapter 8

1. A, B, C. A good SDK is easy to use, well documented, integrated well with other SDKs, has a minimal impact on hardware resources, and provides value-added functionality.
2. A, B. Some of the advantages of using an SDK are quicker integration, faster and more efficient development, brand control, increased security, and the availability of metrics.
3. A, B. The Cisco Meraki cloud platform provides the following APIs to developers: Captive Portal API, Scanning API, MV Sense Camera API, and Dashboard API.
4. A. The Cisco Meraki Dashboard API authentication header is called X-Cisco-Meraki-API-Key.
5. A. The base URL for the Cisco Meraki Dashboard API is https://api.meraki.com/api/v0.
6. C. Cisco Catalyst Center API authentication is based on basic auth.
7. A. The **timestamp** parameter in Cisco Catalyst Center APIs is in UNIX epoch time, in milliseconds.
8. B. Cisco Catalyst Center allows customers to have their non-Cisco devices managed by Catalyst Center through a multivendor SDK. Cisco Catalyst Center communicates with the third-party devices through device packages. The device

packages are developed using the multivendor SDK and implement southbound interfaces based on the CLI, SNMP, or NETCONF.

9. C. Cisco vManage is a centralized network management system that provides a GUI and REST API interface to the SD-WAN fabric.
10. D. The information sent over the authentication POST call is URL form encoded and contains the username and password for the vManage instance.

Chapter 9

1. B. The Cisco Nexus 9000 family of switches can run in two separate modes of operation, depending on the software that is loaded. The first mode is called standalone (or NX-OS) mode, which means the switches act like regular Layer 2/Layer 3 data center devices, which are usually managed individually. The second mode is called ACI mode, in which the Cisco Nexus devices are part of an ACI fabric and are managed in a centralized fashion.
2. B. Bridge domains represent the Layer 2 forwarding domains within the fabric and define the unique MAC address space and flooding domain for broadcast, unknown unicast, and multicast frames. Each bridge domain is associated with only one VRF instance, but a VRF instance can be associated with multiple bridge domains.
3. A. APIC REST API username- and password-based authentication uses a special URI that includes aaaLogin, aaaLogout, and aaaRefresh as the DN targets of a POST operation.
4. B. The service profile is a logical construct in UCS Manager that contains the complete configuration of a physical server. All the elements of a server configuration—including RAID levels, BIOS settings, firmware revisions and settings, adapter settings, network and storage settings, and data center connectivity—are included in the service profile.
5. A. The Cisco UCS Python module for UCS Manager is called ucsmsdk. It can be installed using pip by issuing the following command at the command prompt: **pip install ucsmsdk**.
6. C. Cisco UCS Manager provides a managed object browser called Visore. Visore can be accessed by navigating to https://<UCS-Manager-IP>/visore.html.
7. D. A workflow is a series of tasks arranged to automate a complex operation. The simplest workflow contains a single task, but workflows can contain any number of tasks.
8. B. Each REST API request must be associated with an HTTP header called X-Cloupia-Request-Key, with its value set to the REST API access key.
9. C. The Intersight API is a programmatic interface to the Management Information Model similar to Cisco ACI and Cisco UCS Manager. Just like Cisco ACI and Cisco UCS Manager, the Cisco Intersight Management Information Model is comprised of managed objects.
10. A. An Intersight API key is composed of a keyId and a keySecret. The API client uses the API key to cryptographically sign each HTTP request sent to the Intersight web service.

Chapter 10

1. A, B, C. Cisco's collaboration portfolio allows video calling, integration of bots, Remote Expert use cases.
2. A, B, D. Teams allows users, third-party apps, and bots to interact with its APIs.
3. C. A JWT token is generated using the guest issuer ID and secret.
4. A. Bots use webhooks to handle events.
5. B. Finesse has a rich API set, and the desktop application is completely built using APIs.
6. B. The Finesse Notification Service sends XMPP over BOSH messages to agents that are subscribed to certain XMPP nodes.
7. A. SetMeeting lets you modify the attributes of a meeting after the meeting has been created.
8. D. xAPI allows developers to programmatically invoke commands and query the status of devices that run collaboration endpoint software or Webex RoomOS software.
9. A, C, D. xAPI does not work with FTP.
10. B. The Administration XML (AXL) API provides a mechanism for inserting, retrieving, updating, and removing data from Cisco Unified Communications Manager.

Chapter 11

1. A. Visibility, segmentation, and protection are the three focus areas of Cisco's security portfolio.
2. B. The Investigate API provides enrichment of security events with intelligence to SIEM or other security visibility tools.
3. C. The Umbrella Enforcement API involves an HTTP POST request, which internally comprises the Investigate API to check whether the domain is safe.
4. A. The response header contains the token **X-auth-access-token**, which needs to be used in all subsequent API calls.
5. B. A named object is a reusable configuration that associates a name with a value.
6. C. APTs allow bad actors to gain access to and control endpoint resources over an extended period to steal valuable data without being detected.
7. A, B, C. The Secure Endpoint API sends events to third-party tools, finds where a file has been seen or executed, and creates groups and manages file lists. Blocking DNS requests at the resolver is an Umbrella capability.
8. B. ISE enables devices and users to be identified and provisioned and enables policies to be applied.
9. B. Cisco Secure Malware Analytics is a unified malware analysis and threat intelligence platform.
10. B. IOCs are used to indicate that the system has been affected by some form of malware.

Chapter 12

1. A, B, D. There are three standards-based programmable interfaces for operating on the YANG data models: NETCONF, RESTCONF, and gRPC.
2. B. By default, the NETCONF server on the device runs on TCP port 830 and uses the SSH process for transport.
3. D. Messages sent with NETCONF use remote procedure calls (RPCs), a standard framework for clients to send a request to a server to perform an action and return the results.
4. C. YANG defines a set of built-in types and has a mechanism through which additional types can be defined. There are more than 20 base types, including binary, enumeration, and empty. Percent is not a built-in data type.
5. C. The YANG header contains the namespace for the module. The namespace is used to uniquely identify each module on a system that implements NETCONF.
6. B. One popular NETCONF client is the Python 3 ncclient library.
7. A, D. RESTCONF data is encoded with either XML or JSON. Compared with NETCONF, RESTCONF has added support for JSON encoding.
8. A. The PATCH method provides a resource patching mechanism. It is equivalent to the NETCONF <edit-config> operation with operation=merge.
9. C. Per the RESTCONF standard, devices implementing the RESTCONF protocol should expose a resource called /.well-known/host-meta to enable discovery of root programmatically.
10. B, C. There are two types of telemetry subscriptions. With a dynamic subscription, the subscriber sends a request, usually via the **ietf-yangpush.yang** data model. A configured subscription is configured via the CLI, NETCONF, or RESTCONF and is persistent between reboots.

Chapter 13

1. D. A SaaS provider offers software for use and maintains all aspects of it. You may be allowed to customize parts of the software configuration, but typically you are not allowed to change anything regarding how the software functions.
2. B. IoT is heavily reliant on sensors detecting and responding to events in real time. The edge computing model allows for centralized control of IoT sensors and real-time response because computing capabilities are closer to the sensor.
3. B. There is no single correct deployment method. Bare-metal, virtualized, cloud-native, containerized, and serverless each fit different workloads, so the workload drives the choice.
4. B. Bare-metal puts the application directly on the server with no hypervisor or container layer in between. Big data workloads benefit most from that dedicated access to CPU, memory, and disk.
5. C. Virtualization separated the application and its operating system from the physical server. Once a workload is just a file, you can copy it, move it, and restart it somewhere else.

6. B. A cloud-native application is built from small, independent services spread across many hosts. Losing one server reduces capacity instead of taking the application down.
7. A, C. Containers differ from virtual machines in that they are lighter (fewer resources used in storage), and they can start as fast as an application, whereas a virtual machine requires the guest operating system to boot so takes more time to start.
8. A. Serverless deployment is great for applications that process data periodically but not designed for continuous use.
9. D. The second way of DevOps is the feedback loop for continuous improvement.
10. C. Continuous integration refers to the merging of development work with a shared code base to facilitate automatic testing and software building.
11. A. A Docker image is built from stacked, read-only layers, and a running container adds a writable layer on top. Docker manages those layers through a storage driver.
12. C. To launch a nginx container on port 80, you run the command **docker container run -p 80:80 -d nginx**.
13. C. API keys, database passwords, and other secrets do not belong in source control or in a container image. Keep them in a secrets manager or inject them at runtime as environment variables.
14. A, C. A reverse proxy sits in front of the application and commonly terminates TLS and handles request routing and caching. Database queries and code compilation are not proxy functions.
15. B. A05:2025 Injection covers untrusted input that reaches an interpreter, which is what both SQL injection and cross-site scripting do.
16. A. NIST SP 800-145 defines cloud computing with five essential characteristics, three service models, and four deployment models.

Chapter 14

1. B. A vulnerability is a weakness or gap in protection efforts. It can be exploited by threats to gain unauthorized access to an asset.
2. B, D. A man-in-the-middle attack allows devices to receive traffic as it flows in the network; a brute-force attack involves using trial and error to crack passwords.
3. A. Penetration (pen) testing is commonly used to find weak spots.
4. B. Nmap is the Network Mapper tool, which is used for network discovery and security auditing.
5. B. MFA uses at least two identity components to authenticate a user's identity.
6. A, C. A one-way hash is used for fingerprinting data.
7. A. Data needs to be secured in multiple locations: while it is in motion (network), at rest (storage), and while in use (memory).
8. B. An IDS is passive, as it receives a copy of a packet, whereas an IPS is active, working on live traffic.

Chapter 15

1. B, D. Historically, network devices were managed through command-line interfaces (CLIs) using protocols such as Telnet and Secure Shell (SSH).
2. C. We've seen in Chapter 8 an example of a network controller with Cisco Catalyst Center. Cisco Catalyst Center can be used to completely configure, manage, and monitor networks.
3. B, C. There are usually two types of approaches to infrastructure as code: declarative and imperative. With the declarative approach, the desired state of the system is defined, and then the system executes all the steps that need to happen in order to attain the desired state. The imperative approach defines a set of commands that have to be executed in a certain order for the system to achieve the desired state.
4. A, C. Implementing infrastructure as code processes leads to shorter deployment times for new infrastructure and faster and easier troubleshooting steps.
5. D. A key component of any CI pipeline is the build server, which reacts to developers committing their code into the central repository and starts the initial tests on the new code features.
6. B. The four steps of a CI/CD pipeline are source, build, test, and deploy.
7. B. Ansible playbooks can be run from a terminal with the **ansible-playbook** command.
8. B. Terraform configuration files typically use HCL, or Hashicorp Configuration Language,

9. B. The **terraform plan** command previews proposed infrastructure changes.
10. B. The NSO Device Manager manages network devices using YANG data models and NETCONF. For devices that natively implement NETCONF and YANG models, the device manager is automatic, and devices that do not support NETCONF are integrated in the platform by using Network Element Drivers (NEDs).
11. C. Network topologies are stored as YAML files and can be easily modified and shared.
12. A, D. The pyATS solution is composed of two main components: the pyATS test framework and the pyATS library, which used to be called Genie but was renamed in an effort to simplify the nomenclature of the product.

Chapter 16

1. B. The transport layer, as the name suggests, is responsible for end-to-end transport of data from the source to the destination of the data traffic. Connection-oriented protocols at the transport layer establish an end-to-end connection between the sender and the receiver, keep track of all the segments that are being transmitted, and have a retransmission mechanism in place.
2. C. The Internet layer in the TCP/IP reference model corresponds in functions and characteristics to the network layer in the OSI model.
3. B, D. There are a large number of application layer protocols, including the Hypertext Transfer Protocol (HTTP), which is used for transferring web pages between web browsers and web servers, and File Transfer Protocol (FTP), which is used for transferring files between a client and a server.
4. D. The TCP/IP reference model transport layer PDU is called a *segment*.
5. D. The Preamble field consists of 8 bytes of alternating 1s and 0s that are used to synchronize the signals of the sender and receiver.
6. B. A MAC address has 48 bits organized as 12 hexadecimal numbers.
7. D. If a switch doesn't have the destination MAC address of the frame, it floods the frame over all the ports except the port on which it was received. As the frame gets flooded throughout the network, eventually it will get to its destination.
8. A. The bit pattern for the first byte in a Class C IPv4 address is 110xxxxx.
9. C. The broadcast address for the network 192.168.0.96/27 is 192.168.0.127.
10. A, B, D. An IPv6 address is 128 bits long, with colons separating entries (x:x:x:x:x:x:x:x, where x is a 16-bit hexadecimal field). Successive fields of zero can be represented as :: but only once per address, and the hexadecimal characters are not case sensitive.

Chapter 17

1. A. Ethernet is a star topology that used concentrators to connect all the nodes in the network.
2. C. A WAN covers a large geographic area and uses public networks.
3. B. With a hub, every frame shows up at every device attached to a hub, which can be really harmful in terms of privacy.
4. C. Multiple VLANs can be implemented with Layer 3 switches, so multiple broadcast can be supported.
5. C. Virtual local-area networks (VLANs) divide a single physical switch into multiple logical networks.
6. B. CEF switching uses FIB tables in order to make switching very fast.
7. B. NAT is used for mapping private IP addresses to public IP addresses.
8. B. The management plane is concerned with administrative access to a network device.
9. B. vSmart is the main brain of an SD-WAN solution and manages the control plane.

Chapter 18

1. A, D. Some of the benefits of using DHCP instead of manual configurations are centralized management of network parameters configuration and reduced network endpoint configuration tasks and costs.
2. C. The DNS recursive resolver is a server that receives DNS queries from client machines and makes additional requests in order to resolve a client query.
3. A, B, E. The benefits of Network Address Translation include conserving public IPv4 address space, additional security due to hiding the addressing for internal networks, and flexibility and cost savings when changing ISPs.

4. A. With SNMPv3, the focus was on security, and additional features such as authentication, encryption, and message integrity were added to the protocol specification.

5. D. The main role of NTP is to synchronize the time on a network of devices. The IETF developed NTP, and the latest version of the protocol, version 4, is defined in RFC 5905. It uses User Datagram Protocol (UDP) at the transport layer, and port 123 is reserved for NTP.

6. B. Layer 3 network diagrams capture all the network information available at Layer 3. All devices that operate at Layer 3—routers, firewalls, load balancers, and so on—and how they are interconnected should be included in these diagrams.

7. C. If DNS is malfunctioning for some reason, end-to-end connectivity is impacted. Network tools such as **nslookup** can be used to troubleshoot DNS functionality.

Appendix B. *CCNA Automation CCNAAUTO 200-901 Official Cert Guide* Exam Updates

Over time, reader feedback allows Pearson to gauge which topics give our readers the most problems when taking the exams. To assist readers with those topics, the authors create new materials clarifying and expanding on those troublesome exam topics. As mentioned in the Introduction, the additional content about the exam is contained in a PDF on this book's companion website, at http://www.ciscopress.com/title/9780135368084.

This appendix is intended to provide you with updated information if Cisco makes minor modifications to the exam upon which this book is based. When Cisco releases an entirely new exam, the changes are usually too extensive to provide in a simple updated appendix. In those cases, you might need to consult the new edition of the book for the updated content. This appendix attempts to fill the void that occurs with any print book. In particular, this appendix does the following:

- Mentions technical items that might not have been mentioned elsewhere in the book
- Covers new topics if Cisco adds new content to the exam over time
- Provides a way to get up-to-the-minute current information about content for the exam

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>
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Technical Content

The current Version 1.0 of this appendix does not contain additional technical coverage.